PCT

WORLD INTELLECTUAL PROPERTY ORGANIZATION
International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification 7 : H04M 7/00

A1

(11) International Publication Number: WO 00/69156

(43) International Publication Date: 16 November 2000 (16.11.00)

(21) International Application Number: PCT/US00/13247

(22) International Filing Date: 12 May 2000 (12.05.00)

(30) Priority Data:
60/133,789 12 May 1999 (12.05.99) US

(71) Applicant: STARVOX, INC. [US/US]; 2125 Zanker Road, San Jose, CA 95131 (US).

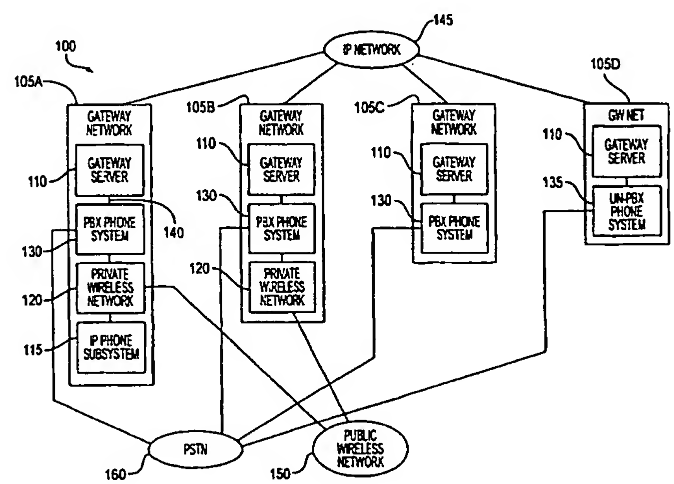
(72) Inventors: DUFFY, Judith; 764 Anacapa Court, Milpitas, CA 95053 (US). RAAD, Stephen, R.; 738 Marin Drive, Mill Valley, CA 94941 (US). CHANG, Gordon, K.; 2954 Heidi Drive, San Jose, CA 95132 (US). BARRY, Richard, B.; 305 South Gordon Way, Los Altos, CA 94022 (US).

(74) Agents: NUTTLE, William, E. et al.; Flehr Hohbach Test Albritton & Herbert LLP, Suite 3400, 4 Embarcadero Center, San Francisco, CA 94111-4187 (US).

(81) Designated States: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BY, CA, CH, CN, CR, CU, CZ, DE, DK, DM, DZ, EE, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, NO, NZ, PL, PT, RO, RU, SD, SE, SG, SI, SK, SL, TJ, TM, TR, TT, TZ, UA, UG, UZ, VN, YU, ZA, ZW, ARIPO patent (GH, GM, KE, LS, MW, SD, SL, SZ, TZ, UG, ZW), Eurasian patent (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE), OAPI patent (BF, BJ, CF, CG, CI, CM, GA, GN, GW, ML, MR, NE, SN, TD, TG).

Published
With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: METHOD AND APPARATUS FOR INTEGRATED VOICE GATEWAY WITH INTERFACE TO MOBILE TELEPHONE, IP TELEPHONE AND UN-PBX SYSTEMS



(57) Abstract

A communication system (100) and a method for operating the same are described to provide seamless, automatic routing of telephone calls over a public switched telephone network (PSTN 160), an internet protocol (IP) network (145), a public-wireless-network (150) and a private-wireless-network (120). In one embodiment, the system (100) comprises a plurality of gateway networks (105) coupled to the PSTN (160), IP network (145) and the public-wireless-network (150). The gateway networks (105) are configured to automatically select over which of the IP network (145), PSTN (160) or the public-wireless-network (150) to route the telephone call. Preferably, the gateway networks (105) are configured to reroute an in-progress telephone call over the IP network (145) over the PSTN (160) if a delay in transmission of date packets, losses in transmission of data packets, or jitter exceeds a specified maximum. More preferably, the gateway networks (105) are configured so that the routing of the telephone call is substantially transparent to the calling party and to the called party.

BEST AVAILABLE COPY

FOR THE PURPOSES OF INFORMATION ONLY

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AL	Albania	ES	Spain	LS	Lesotho	SI	Slovenia
AM	Armenia	FI	Finland	LT	Lithuania	SK	Slovakia
AT	Austria	FR	France	LU	Luxembourg	SN	Senegal
AU	Australia	GA	Gabon	LV	Latvia	SZ	Swaziland
AZ	Azerbaijan	GB	United Kingdom	MC	Monaco	TD	Chad
BA	Bosnia and Herzegovina	GE	Georgia	MD	Republic of Moldova	TG	Togo
BB	Barbados	GH	Ghana	MG	Madagascar	TJ	Tajikistan
BE	Belgium	GN	Guinea	MK	The former Yugoslav Republic of Macedonia	TM	Turkmenistan
BF	Burkina Faso	GR	Greece			TR	Turkey
BG	Bulgaria	HU	Hungary	ML	Mali	TT	Trinidad and Tobago
BJ	Benin	IE	Ireland	MN	Mongolia	UA	Ukraine
BR	Brazil	IL	Israel	MR	Mauritania	UG	Uganda
BY	Belarus	IS	Iceland	MW	Malawi	US	United States of America
CA	Canada	IT	Italy	MX	Mexico	UZ	Uzbekistan
CF	Central African Republic	JP	Japan	NE	Niger	VN	Viet Nam
CG	Congo	KE	Kenya	NL	Netherlands	YU	Yugoslavia
CH	Switzerland	KG	Kyrgyzstan	NO	Norway	ZW	Zimbabwe
CI	Côte d'Ivoire	KP	Democratic People's Republic of Korea	NZ	New Zealand		
CM	Cameroon			PL	Poland		
CN	China	KR	Republic of Korea	PT	Portugal		
CU	Cuba	KZ	Kazakstan	RO	Romania		
CZ	Czech Republic	LC	Saint Lucia	RU	Russian Federation		
DE	Germany	LI	Liechtenstein	SD	Sudan		
DK	Denmark	LK	Sri Lanka	SE	Sweden		
EE	Estonia	LR	Liberia	SG	Singapore		

METHOD AND APPARATUS FOR INTEGRATED VOICE GATEWAY WITH INTERFACE TO MOBILE TELEPHONE, IP TELEPHONE AND UN-PBX SYSTEMS

Cross Reference to Related Applications

This application claims priority from United States Provisional Patent Application Serial Number 60/133,789 filed May 12, 1999 which is incorporated herein by reference.

Field of the Invention

This application relates generally to telephony systems, and more particularly to an integrated voice gateway system with a wireless capability and a method of operating the same.

Background

The widespread popularity of the Internet has provided new means of rapid and comprehensive communication between users located in distant and diverse locations around the world. Methods of sending, finding and retrieving information, previously confined to the domain of government, academia and industry, are now available in business, in the community, and in the home. Formerly arcane technical terms such as Telenet, electronic mail (e-mail), file transfer protocol (FTP), hypertext transfer protocol (HTTP) and world wide web (WWW or web) are now widely used.

Very soon after the popularity of the Internet became widespread, new applications of the underlying technology began to emerge. With the concomitant growth of multimedia, a predominately text-based medium quickly expanded to include graphics, imagery, motion pictures and sound. A natural extension of the capability to transmit

recorded, digitized sound between personal computers (PC), was the advent of PC based telephony. Although the initial users of PC to PC telephone calls over the Internet were primarily computer hobbyists and the like, there was an early recognition of the fact that the Internet provided the potential for the average user to make a telephone call anywhere in the world for the cost of a local telephone call to an Internet service provider (ISP).

PC to PC telephone technology is limited by the need to be logged on to a PC and the Internet to place or receive a call. Software incorporating proprietary algorithms limit the ability to call to others having the same or similar software. The sound quality is often degraded because of packet loss and delays in forwarding packets from the sender to the receiver over the Internet, operation in a half-duplex mode, and the use of low quality PC speakers and microphones.

With the expectation of improved performance and reduced cost of telephone calls in the business environment, voice gateways have facilitated the interconnection of the private branch exchange (PBX) and the computer network. As used herein, PBX includes hybrid, key systems, and other such systems. Thus, through a PBX coupled to an Internet protocol (IP) network (e.g., intranet, wide area network (WAN), Internet), telephone calls between different sites within a company, or other institution, organization or enterprise (hereinafter referred to as "company"), or between companies, the company or companies having installations at two or more locations which locations may be geographically distant from each other, may be routed over the IP network rather than via the public switched telephone network (PSTN). As used herein, the PSTN includes both public and private networks. This can result in significant cost savings and can also help to improve communication within and between companies by providing a variety of related services which are not available via the PSTN.

The level of integration achieved in current voice gateway systems is quite low, and such systems are limited in the services they can provide. In particular, current voice gateway systems are capable of only routing a nominal telephone call from a calling party at point A to a called party at point B. However, if, for example, the called party is not present, or if the called party's telephone is currently busy, current voice gateway systems do not provide important additional services to facilitate making a connection between the calling party and the called party at a later time or at another location or by an alternative method.

One of the reasons for the limitations is that current voice gateway systems are limited in their ability to obtain, store, update and retrieve necessary information about both the calling party and the called party in order to do anything other than simply attempt to make a straight forward connection between the two points. If the telephone system had sufficient information about both parties, then the system could facilitate making the connection at a later time, at another location or by an alternative method. However, in current voice gateway systems, there is no way to obtain the necessary call status and call control information, nor is there an accessible central data base in which to store and from which to retrieve this information. Current voice gateway systems have no real-time call control/call status information link with the PBX, nor do they have any storage of telephone user information. For example, current voice gateway networks have no information regarding the calling party's name, telephone number, or status of the called party, e.g., busy or idle. It is this information about the calling and called parties which is not readily available, but which is necessary to provide important additional services.

Accordingly, there is a need for a highly integrated voice gateway system for use within a company and between companies having installations at two or more locations which locations may be geographically distant from each other. The integrated voice gateway system should have the ability to route telephone calls between parties at two different locations over the IP network, the PSTN, a private-wireless-network and a public-wireless-network, and the capability to automatically select over which to route telephone calls.

Summary of the invention

The present invention overcomes the disadvantages of the prior art by providing a communication network having an integrated voice gateway system incorporating a private-wireless-network as a component of the system and method for operating the same that provides users of mobile phones and un-PBX systems access to features of the gateway network as described in U.S. Pat. App. Serial No. 09/061,802, which is incorporated herein by reference.

In one aspect, the present invention is directed to a communication system for providing communication between a plurality of sites within an enterprise. The

communication system comprises a public switched telephone network (PSTN), an internet protocol (IP) network, a public-wireless-network and a plurality of gateway networks coupled to the PSTN, IP network and the public-wireless-network to route a telephone call between a calling and a called party thereover. Each of the plurality of gateway networks is configured to automatically select over which of the IP network, PSTN or the public-wireless-network to route the telephone call. Preferably, the plurality of gateway networks are configured to automatically reroute an in-progress telephone call routed over the IP network over the PSTN if a delay in transmission of data packets, losses in transmission of data packets, or jitter exceeds a specified maximum. More preferably, the gateway networks are configured so that the rerouting of the in-progress telephone call is substantially transparent to the calling party and to the called party.

In one embodiment, at least one of the plurality of gateway networks comprises a private-wireless-network, and the gateway network is configured to route a telephone call between the calling and the called party over the private-wireless-network and at least one of the IP network, the PSTN or the public-wireless-network.

In another embodiment, one of the gateway networks comprises an UN-PBX system, and wherein the gateway network is configured to route a telephone call between the calling and the called party over the UN-PBX system.

In yet another embodiment, when a called party's telephone is unavailable, or the called party does not answer, the gateway networks are configured to automatically set up a telephone call between the calling party and the called party as soon as the called party becomes available. In one version of this embodiment, the gateway networks are configured to enable the calling party to forward the telephone call. Optionally, the gateway networks are configured to create a log of incoming telephone calls, call attempts and outgoing telephone calls. The gateway networks may also be configured to enable a user of the system to forward telephone calls to a different telephone according to a time schedule predetermined by the user.Alternatively, the gateway networks are configured to enable the calling party to send a computer message that will be immediately displayed on a computer screen co-located with the called party's telephone.

In still another embodiment, at least one of the plurality of gateway networks is coupled to a directory server comprising information on parties using the communication system, and wherein the gateway network is configured to (i) provide a single-point-of-

entry for modifications to the information and (ii) to provide replication of these changes across all enterprise sites. In one version of this embodiment, the gateway networks are further configured to provide information identifying the calling party. The identifying information can be displayed on a display of a called party's telephone or on a computer screen co-located with the called party's telephone.

In another aspect, the present invention provides a method of operating a communication system to connect a telephone call between a calling party on a first telephone and a called party on a second telephone. In the method, a plurality of gateway networks are coupled to one another via a public switched telephone network (PSTN), an internet protocol (IP) network and a wireless-network. The communication system automatically selects over which of the IP network, PSTN or the wireless-network to setup the telephone call, and routs the telephone call over at least one of the IP network, PSTN or the wireless-network.

In one embodiment, at least one of the plurality of gateway networks comprises a private-wireless-network, and the step of routing the telephone call between the calling and the called party comprises the step of routing the telephone call over the private-wireless-network and at least one of the IP network, the PSTN or the wireless-network.

In another embodiment, at least one of the first and second telephones is a mobile phone coupled to the private-wireless-network , and the step of automatically selecting over which of the IP network, PSTN or the wireless-network to setup the telephone call includes the step of determining whether the mobile phone is turned on.

In yet another embodiment, the wireless-network is a public-wireless-network and at least one of the first and second telephones is a mobile phone coupled to the public-wireless-network, and the step of automatically selecting over which of the IP network, PSTN or the wireless-network to setup the telephone call includes the step of learning whether the mobile phone in public-wireless-network is turned on.

In still another embodiment, the method comprises the further steps of automatically rerouting an in-progress telephone call routed over a first call path over the public-wireless-network to a second call path over the private-wireless-network when the mobile phone needs to roam off the public-wireless-network to the private-wireless-network.

It is the object of the invention to provide an integrated voice gateway system with a wireless private network component for use within a company or enterprise. This gateway system can route a voice telephone call between parties at two different locations over a wide area network (WAN) IP network, where either or both parties may be using a mobile phone. It is a further object of the invention that the gateway system will automatically select which of the IP network or PSTN over which to route the calls.

It is a further object of the invention to provide a system which can route a telephone call between a calling party using a mobile phone at a first location within the system to a second location within the system via the WAN IP network, and then from the second location to a called party at a third location via the PSTN.

It is an object of the invention to provide an integrated voice gateway system that can place a telephone call over an IP network and if, during the telephone call the quality of the telephone call falls below a predetermined quality level, reroute the telephone call over to the PSTN. Preferably, the integrated voice gateway system is able to do so in a manner that is transparent to both the calling and called parties, either or both of which can be using a mobile phone.

It is an object of the invention to provide an integrated voice gateway system that can track any modifications to any telephone-user-profile in the enterprise. Telephone-users include users of PBX telephones and mobile phones, and modifications to the user-profile may include changes to the existing data for a user, addition of new users or deletion of users. It is a further object of the invention to provide an integrated voice gateway system that can integrate with an enterprise-directory to allow single point of entry for user profile modifications and to provide replication of these changes across all enterprise sites.

It is an object of the invention to provide an integrated voice gateway system in which the identification of the calling party (e.g., name, title, department, primary telephone number) can be displayed on the called telephone's display, where the called party is using a mobile phone that supports caller ID display. Likewise, the identification of the calling party can be displayed on any other telephone in the integrated voice gateway system, where the calling party is using a mobile phone.

It is an object of the invention to provide an integrated voice gateway system in which the identification of the calling party (e.g., name, title, department, primary

telephone number) is displayed on a computer screen of a desktop workstation or personal computer (PC) (rather than on a telephone display) co-located with the called party's telephone, and that such information be displayed regardless of the vendor(s) supplying the telephone equipment used by the calling and called parties. Either or both of the calling and called parties may be using a mobile phone. It is further an object of the invention that such information be provided via a WWW browser interface regardless of the vendor(s) supplying the desktop workstation or PC or an operating system used thereon.

It is an object of the invention to provide an integrated voice gateway system that can create a log of incoming telephone calls and call attempts arriving over a WAN IP network and of outgoing telephone calls and call attempts routed over the WAN IP network and identify the names of calling and called parties. It is a further object of the invention to provide a log of all incoming and outgoing calls whether the calling and called parties are using a PBX telephone, a PSTN telephone, or a mobile phone.

It is an object of the invention to provide an integrated voice gateway system in which, when a called party's telephone is busy, the system can automatically set up a call between the calling party and the called party as soon as the called party hangs up. It is a further object of the invention to provide such a capability even when either or both of the calling party and called party are using a mobile phone. It is yet a further object of the invention to provide such a capability even when a called party has voicemail feature available.

It is an object of the invention to provide an integrated voice gateway system in which, when a called party is busy, the calling party may send a computer message that will be immediately displayed on a computer screen co-located with the called party's telephone, for example to explain why the calling party needs to speak with the called party. It is a further object of the invention to provide such a capability even when either or both of calling party and called party are using a mobile phone.

It is an object of the invention to provide an integrated voice gateway system in which, when a called party does not answer an incoming telephone call, the calling party may forward the call, for example to voicemail, or to an answering station (e.g., a receptionist or other designated party). It is a further object of the invention to provide such a capability even when calling party and/or the called party are using a mobile

phone. It is yet a further object of the invention to provide the capability for a party at an answering station to send a computer message which will immediately be displayed on a computer screen co-located with the called party's telephone.

It is an object of the invention to provide an integrated voice gateway system in which a user of the system may set up the system to forward that user's telephone calls to a different telephone. It is a further object of the invention to forward calls to a PSTN telephone, a cellular telephone in a public-wireless-network, a PC-based IP telephone or a mobile phone. It is a further object of the invention to provide the capability for a user to set up the system to forward that user's telephone calls to different telephones according to a time schedule predetermined by the user. It is a still further object of the invention to provide the capability for a user to set up the system to forward telephone calls originating only from one or more calling parties or telephone numbers so designated by the user. It is a further object of the invention to provide the capability to set up call forwarding via a browser interface or interactive voice response (IVR).

Brief Description of the Drawings

These and various other features and advantages of the present invention will be apparent upon reading of the following detailed description in conjunction with the accompanying drawings, where:

FIG. 1 is a block diagram of a communication network having gateway networks according to an embodiment of the present invention;

FIG. 2 is a block diagram of a gateway network according to an embodiment of the present invention;

FIG. 3 is a block diagram of a gateway server shown FIG. 2 configured to operate with a PBX according to an embodiment of the present invention;

FIG. 4 is a block diagram of a gateway server shown FIG. 2 configured to operate with an Un-PBX according to an embodiment of the present invention;

FIG. 5 is a block diagram of a private-wireless-network shown FIG. 2 according to an embodiment of the present invention;

FIG. 6 is a block diagram of an IP telephone network shown FIG. 2 according to an embodiment of the present invention;

FIG. 7 illustrates operation of a gateway network according to an embodiment of

the present invention to provide a connection between a mobile phone operating in the private-wireless-network and a PSTN telephone at a nearby location;

FIG. 8 illustrates an embodiment of a mobile-to-PSTN call setup sequence for the gateway network of FIG. 7;

FIG. 9 illustrates an embodiment of a PSTN-to-mobile call setup sequence for the gateway network of FIG. 7;

FIG. 10 illustrates an embodiment of a mobile-to-PBX call in progress for a telephone call routed within a single gateway network;

FIG. 11 illustrates an embodiment of a mobile-to-PBX call setup sequence for the telephone call of FIG. 10;

FIG. 12 illustrates an embodiment of a mobile-to-mobile call in progress for a telephone call routed within a single gateway network;

FIG. 13 illustrates an embodiment of a mobile-to-mobile call setup sequence for the telephone call of FIG. 12;

FIG. 14 illustrates operation of a gateway network according to an embodiment of the present invention to provide a mobile-to-PSTN telephone call, WAN VoIP and Hop-off to PSTN;

FIG. 15 illustrates operation of a gateway network according to an embodiment of the present invention to provide WAN VoIP for an in-progress mobile-to-PBX telephone call originating from a mobile phone;

FIG. 16 illustrates an embodiment of a mobile-to-PBX call setup sequence for the telephone call of FIG. 15;

FIG. 17 illustrates operation of a communication network according to an embodiment of the present invention to provide an in-progress mobile-to-mobile phone call for mobile phones coupled to different gateway networks;

FIG. 18 illustrates operation of a communication network according to an embodiment of the present invention to provide a telephone call for a mobile phone in private-wireless-network;

FIG. 19 illustrates operation of a communication network according to an embodiment of the present invention to provide a telephone call for a mobile phone in public-wireless-network;

FIG. 20 illustrates operation of a communication network according to an

BNSDOCID: <WO 0069156A1 I >

embodiment of the present invention to provide a telephone call between a mobile phone in a public-wireless-network and a mobile phone homed at a remote site;

FIG. 21 illustrates operation of a communication network according to an embodiment of the present invention to provide an in-progress mobile-to-PBX telephone call for a mobile phone in a public-wireless-network calling to local PBX telephone;

FIG. 22 illustrates an embodiment of a mobile-to-PBX call setup sequence for the telephone call of FIG. 21;

FIG. 23 illustrates operation of a communication network according to an embodiment of the present invention to provide an in-progress PBX-to- mobile phone call for a PBX telephone calling to a mobile phone in a public-wireless-network;

FIG. 24 illustrates an embodiment of a PBX-to-mobile call setup sequence for the telephone call of FIG. 23, wherein the mobile phone is homed locally;

FIG. 25 illustrates an embodiment of a PBX-to-mobile call setup sequence for the telephone call of FIG. 23, wherein the mobile phone is homed remotely;

FIG. 26 illustrates operation of a gateway network according to an embodiment of the present invention to provide a connection between a PBX telephone and a mobile phone operating in a public-wireless-network at a remote site;

FIG. 27 illustrates operation of a gateway network according to an embodiment of the present invention to provide a fallback-to-PSTN for an in-progress mobile-to-PBX telephone call originating from a mobile phone;

FIG. 28 illustrates operation of a gateway network according to an embodiment of the present invention to provide an in-progress PBX-to-mobile phone call for a PBX telephone calling to a mobile phone in a private-wireless-network of the same gateway server;

FIG. 29 illustrates a call path for the telephone call of FIG. 28 after mobile phone user's action to transfer the telephone call to another user within the enterprise;

FIG. 30 illustrates an embodiment of a transfer sequence for the telephone call of FIG. 28;

FIG. 31 illustrates operation of a gateway network according to an embodiment of the present invention to provide a three-way conference-call;

FIG. 32 illustrates operation of a gateway network according to an embodiment of the present invention to provide dial call control via a PC browser; and

FIG. 33 illustrates operation of a gateway network according to an embodiment of the present invention to provide busy notification and alert pop-up.

Detailed Description of the Invention

An apparatus and method are provided for integrating a voice gateway with wireless telephone systems, internet protocol (IP) telephone systems, and un-PBX telephone systems to provide voice over internet protocol (VoIP) capability and feature transparency to users at geographically distinct sites. By un-PBX system it is meant a dedicated or general purpose computer comprising a telephone interface and a software application that is capable of providing many features of a traditional private branch exchange (PBX).

DIFFERENT CONFIGURATIONS OF THE GATEWAY NETWORK

FIG. 1 shows a sample configuration of a communication system or network 100 for multi-site company having four gateway networks 105 (individually 105A, 105B, 105C and 105D) at a different company site. Each gateway networks 105 comprises a gateway server 110, and one or more of an IP telephone subsystem 115, a private-wireless-network 120 and a private branch exchange (PBX) system 130 or an un-PBX telephone system 135 connected to each other by a local area network (LAN) 140. Each gateway network 105 can communicate to other gateway networks to send voice and data to each over an IP Network 145, a public-wireless-network 150 or the public switched telephone network (PSTN) 160.

In the gateway network 105, the gateway server 110 operates in conjunction with the PBX telephone system 130 to coordinate both PBX telephone calls and calls over the PSTN 160. The configuration of the gateway network 105 is described in greater detail in U.S. Pat. App. Serial No. 09/061,802, which is incorporated herein by reference.

The private-wireless-network 120 supports mobile phone users using private mobile phones. By private mobile phones it is meant wireless phones that will operate with the private-wireless-network 120 and, preferably, also with the public-wireless-network 150. The gateway server 110 coordinates the telephone-related activities for both PBX telephones (not shown in this figure) in the PBX telephone system 130 and mobile phones (not shown in this figure) in the private-wireless-network 120, as well as for calls

incoming from and outgoing to the PSTN 160. Furthermore, the private-wireless-network 120 can have a connection with the public-wireless-network 150, to allow for roaming between the private and public-wireless-networks.

The IP telephone subsystem 115 supports IP telephones that can be discrete, individual IP telephones (not shown) or can be incorporated into computer work stations (not shown). The gateway server 110 coordinates the activities of all internal, company telephone systems including the PBX telephone system 130, the private-wireless-network 120 and the IP telephone subsystem 115. PSTN calls and the activities of roaming mobile phones are also coordinated by the gateway server 110.

The gateway network 105, can also include an un-PBX telephone system 135. An "un-PBX" telephone system is a cheaper alternative to a traditional PBX system. The un-PBX system consists of a software application plus a set of telephony cards which can be plugged into a standard PC (not shown), such as a Windows NT workstation. Because of its' relatively low cost, un-PBX telephone systems 135 have become a popular alternative for small offices.

It is to be understood that the possible gateway network configurations that can be supported by the invention are not limited to those shown in FIG. 1. The table below shows some combinations of network telephone system elements that can be supported by the invention. Furthermore, for any kind of telephone system that can be supported, more than one of that kind of system can be equally well supported by the invention. For example, a configuration consisting of two PBX telephone systems 130 and one private-wireless-network 120 can all be controlled by a single gateway server 110.

TABLE I

Configuration	PBX Phone System	UN-PBX Phone System	Private Wireless Network	IP Phone System
1. Gateway Network 105A	X			
2. Gateway Network 105B	X		X	
3. Gateway Network 105C	X		X	X
4.	X			X
5. Gateway Network 105D		X		
6.		X	X	
7.		X	X	X
8.		X		X
9.			X	
10.			X	X
11.				X

For example, the architecture of the gateway network 105C as shown in FIG. 2 corresponds to the third configuration from the table. The gateway server 110, IP telephone systems 115, and private-wireless-networks 120 are all connected to the LAN 140. The gateway server 110 is coupled with one or more PBX telephone systems 130. Each PBX telephone system 130 is coupled to the PSTN 160. The gateway server 110 is coupled to the IP Network 145 via a router 165. Most routers 165 have configurations which can handle separating the VoIP data versus other data when routing packets incoming from the IP Network 145. The communication system can further include a directory server 170. The directory server 170 may be on the same server as the gateway server 110 or it may be physically located on a different computer (not shown). Directory servers are described in greater detail in U.S. Pat. App. Serial No. 09/061,802.

GATEWAY SERVER DETAILED DESCRIPTION

A simplified view of the gateway server 110 is shown in FIG. 3. For purposes of clarity, many of the details of the gateway server 110, previously described in U.S. Pat. App. Serial No. 09/061,802, have been omitted. The gateway server 110 comprises

gateway server software 175 some key components of which are shown, including a database 180, a web server 190, a directory services module 195 and a computer telephony integration (CTI) services module 200. The database 180 is where configuration and user data is stored. The web server 190 supports web browser and graphical user interface (GUI) applications, described infra. The web server 190 can be any suitable, commercially available web server. The directory services module195 refers to an interface software being used to access the user information. If an external directory server 205, such as an enterprise directory server, is in use and available on the LAN, then the directory services module195 provides the interface function to that server. If no external or enterprise directory server 205 is available, the directory services module 195 provides both the directory server and the interface functions. The CTI module 200 provides an interface to a CTI driver 210, which is typically supplied by the PBX vendor and is specific to the PBX telephone system 130. The gateway server 110 further includes a VoIP driver 220 which consists of hardware and software that handles the conversion between the voice data in the format required for a trunk or station telephony interface and the IP packet format. A trunk driver 225 includes hardware and software to interface the trunk with the PBX. A station driver 240 consists of hardware and software that manages the analog station interface to the PBX telephone system 130.

PBX SYSTEM DETAILED DESCRIPTION

For purposes of discussion in this description, the PBX telephone system 130 consists of a PBX 250 and one or more PBX telephones 255. For purposes of clarity, many of the details of PBX telephone systems 130, PBXs 250 and PBX telephones 255 that are widely known have been omitted from the figures and the description. However, PBX telephone systems 130 and the PBX telephones 255 and how they work are discussed in greater detail in U.S. Pat. App. Serial No. 09/061,802.

UN-PBX TELEPHONE SYSTEM DETAILED DESCRIPTION

FIG. 4 shows a sample architecture for the gateway server 110 integrated with an un-PBX telephone system 135. Conventional un-PBX systems use un-PBX software on a separate or dedicated server platform to control an office's telephone system to provide both person-to-person calls in the office and outside calls. One shortcoming of such a

configuration is that integration with other company sites is not possible. However, with the present invention, the components of the un-PBX telephone system 135 are part of the gateway server 110, enabling integration with other company sites and resulting in a substantial cost savings by eliminating the need for a dedicated server platform for the un-PBX telephone system. The un-PBX telephone system 135 of the present invention includes hardware (not shown) and software 260 to interface with and/or control the trunk and the station. In addition, the un-PBX telephone system 135 can include a basic VoIP hardware/software package to convert the digital IP packets to an analog format compatible with the telephones of the UN-PBX telephone system.

The un-PBX software 260 typically contains PBX switching software 280 that has an open application programming interface (API) (not shown). The gateway server software 175 uses this API to control the call setup of the office telephones in a manner similar to how it controls PBX telephones 255 in a PBX configuration. For some of the more sophisticated features, such as call control, the CTI services module 200 of the gateway server software 175 also communicates with the un-PBX software 260 to coordinate these features.

Users of the un-PBX telephone system 135 can call to other sites using the standard dial plan, i.e., a location ID plus a called party's extension. Operation of the un-PBX telephone system 135 is transparent to the users whether the site being called to has an un-PBX telephone system 135 or PBX telephone system 1320. Thus, the gateway server 110 of the present invention, provides seamless and consistent usage of the communication system for users at all sites and is an attractive alternative to a full PBX configuration for a small satellite office.

Private-wireless-network DETAILED DESCRIPTION

FIG. 5 shows the major components of the private-wireless-network 120 including a mobile phone 285 and a wireless base station 290 (also known as a wireless base transceiver station).

By mobile phone 285 it is meant any a portable communications device such as a cellular telephone or a personal communication service (PCS) telephone. Users can place and receive telephone calls from the mobile phone 285. Users are typically employees at a private company site or campus that carry a mobile phone 285 with them

when they leave their desk to perform company business at other locations or offices within the company campus, or when they visit other company locations. In some implementations, this mobile phone 285 can roam into the public-wireless-network 150.

The wireless base station 290 comprises hardware and software that enable it to receive and transmit voice and data to or from the mobile phones 285. The wireless base station 290 converts the voice data received via the mobile phone 285 to H.323 IP packets and routes them to the gateway server 1110 via the LAN 140. Similarly, H.323 IP packets for voice communication destined for the mobile phone 285 are routed from the gateway server 110 out to the wireless base station 290 via the LAN 140, where the packets are converted back to a form compatible with the mobile phone 285 and transmitted to the mobile phone. Typically, multiple wireless base transceiver stations 290 will be placed on a given campus, each with a defined wireless range. As a mobile phone user travels across campus, the current call is handed off from one wireless base station 290 to another.

A wireless gatekeeper server 295 coordinates control of the wireless base stations 290 and their associated mobile phone calls. The wireless gatekeeper server 295 keeps track of which mobile phones 285 are currently operating (i.e., recently registered or handling an in-progress call) and which wireless base stations 290 each is communicating with. The wireless gatekeeper server 295 communicates to the wireless base stations 290 using H.323 signaling. The wireless gatekeeper server 295 may also retain a list of IDs identifying mobile phones that are permitted to access the wireless service, or it may request this information as needed from the gateway server 110. The wireless gatekeeper server 295 is the primary point within the private-wireless-network 120 for call control interaction with the gateway server 110. The wireless gatekeeper server 295 may control both local wireless base stations 290 and remote wireless base stations 300. The private-wireless-network 120 may be so configured to handle wireless services for employees at a small satellite sales office.

The wireless gatekeeper server 295 comprises software (not shown) which can reside on its own single-purpose server, on the gateway server 110 or on one of the wireless base stations 290. Typically, the wireless gatekeeper server 295 is scalable so that a single one can serve a site, with the wireless gatekeeper server interacting with its associated gateway server 110, the wireless base stations 290, 300. The associated

gateway server 110 and the wireless base stations 290, 300, whose tasks are coordinated by wireless gatekeeper server 295 make up the private-wireless-network 120. However, there may also be multiple wireless gatekeeper servers 295, for example, to serve a large number of mobile phone users. In this configuration, one of the wireless gatekeeper servers 295 may be configured as a master, and the other, slave wireless gatekeeper servers would communicate only with that wireless gatekeeper server. Alternatively, the gateway server 110 would simply communicate to each wireless gatekeeper server 295 separately, and the wireless gatekeeper servers would need to communicate with each other for purposes of handoff only. A handoff is when a mobile phone user moves from the area covered by one wireless base station 295 to the area covered by an adjacent wireless base station 300. "Handoff" is the term used in North America; "handover" is the European term.

The public-wireless-network 150 shown in FIG. 5 is connected with the PSTN 160; the PSTN can route calls into the public-wireless-network, and vice versa. Furthermore, some private-wireless-networks 120 support configurations whereby the private-wireless-network and the public-wireless-network 150 are connected, as shown in FIG. 5, via a signaling interface 310; thus the public-wireless-network 150 can keep the private-wireless-networks 120 apprized regarding which mobile phones 290 are operating in the public-wireless-network. Thus, the communication system 100 of the present invention enables a mobile phone user (using either a private or public mobile phone) to roam back and forth between the two wireless-networks 120, 150. Given the complex security, authentication and billing issues involved, such configurations are rare in conventional communication systems.

The public-wireless-network 150 may include conventional cellular communications standards, including GSM, IS-136, AMPS, TAC, CDMA, and so on. The public-wireless-network 150 may also include other services, such as PCS, for implementing wireless-networks for voice and/or data communication.

IP TELEPHONE DETAILED DESCRIPTION

FIG. 6 shows the IP telephone subsystem 115 which is managed by a centralized server called the IP call manager 315. The call manager 315 may reside on its own server platform (as shown), or it may reside on the same platform as the gateway server 110.

The individuals that use the IP telephone services are equipped with one of (i) an IP phone 320, (ii) an IP telephone and a work station 325, or (iii) a work station that has a built-in IP telephone 330. The IP call manager 315 is the server that first receives a call setup request from an IP telephone user who initiates a call on the IP phone 320. If the call is for an extension representing another IP phone in the IP telephone subsystem 115, the call manager 315 connects the two IP phones so a call can take place. As the connection is set up, the call manager 315 exchanges information with the gateway server 110 including caller ID information (normally stored at the gateway server). In addition, the gateway server 110 can log the start of the call to a call log (not shown). If a call is initiated to an extension or to a location ID plus an extension that is not known to the call manager 315, then the call manager passes the call setup request to the gateway server 110. The gateway server 110 then determines how to route the call, sets up the other leg of the call, and returns to the IP call manager 315 the IP address to which the IP telephone needs to route the IP packets.

CALL ROUTING AND SETUP FOR Private-wireless-network CALLS

There are a number of different paths that must be handled for routing a call involving a mobile phone 285 in the private-wireless-network 120. These different routing paths and the sequence of steps required to establish these routing are described in the next paragraphs.

FIG. 7 MOBILE-TO-PSTN ROUTING SAME GATEWAY NETWORK

In FIG. 7 a call in progress is shown between a mobile phone 285 operating in the private-wireless-network 120 and a PSTN telephone 340 at a location nearby the gateway network's location. Starting with the caller mobile phone 285, the voice data travels to the wireless base station 290 where is it converted into IP packets which are dropped onto the local LAN 140. The packets are then routed to the VoIP driver 220 of the gateway server 110. At the VoIP driver 220, the data is removed from the IP packets and decoded from whatever encoding and compression format that was applied within the wireless private network 120. The voice data is then routed via the trunk driver 225 to the PBX telephone system 130 and from there to the PSTN 160, which in turn routes it in an analog format to the PSTN telephone 340.

The voice which travels from the called party to the caller follows the same path, but goes through the process in reverse; at the VoIP driver 220, the voice data is encoded and inserted into IP packets. The data from the IP packets are extracted and decoded within the private-wireless-network 120.

In FIG. 7, a caller and called telephone have been identified. However, it should be noted that the same routing path would result if the caller where using the PSTN telephone 340 and the called party on the mobile phone 285.

In determining how to setup a call such as that shown in FIG. 7, there are several variations that can be considered each of which will result in different sequences of steps in the call setup process. Some of these variations will now be described in more detail.

FIG. 8 - MOBILE TO PSTN CALL SETUP SEQUENCE

1. User dials and hits SND button on mobile phone 285.
2. Mobile phone 285 communicates with wireless base station 290 to exchange initial setup information.
3. Wireless base station 290 communicates to wireless gatekeeper server 295 that a call setup is requested.
4. Wireless base station 290 verifies the mobile ID for the user is OK to access the private-wireless-network 120.
5. Wireless base station 290 communicates to gateway server 110 that call setup is requested.
6. Gateway server 110 authenticates, via the enterprise directory 205, the mobile ID and user privilege to call to PSTN 160, and verifies that PBX telephone system 130 resources are available to connect to the PSTN.
7. Gateway server 110 communicates call setup request to PBX telephone system 130.
8. PBX telephone system 130 communicates call setup request to PSTN 160 via the station driver 240.
9. PSTN 160 rings called party PSTN telephone 340.
10. Called party answers.

11. PSTN 160 informs PBX telephone system 130 that called party has answered. PBX telephone system switches the routing through the Station/Trunk Driver from the station over to the trunk.

12. PBX telephone system 130 informs gateway server 110 that called party has answered.

13. Gateway server 110 informs the wireless gatekeeper server 295 that called party has answered.

14. The wireless gatekeeper server 295 informs the wireless base station 290 that the called party has answered.

15. The wireless base station 290 then connects with the VoIP Driver 220.

16. Wireless base station 290 coordinates with mobile phone 285 to start routing IP voice packets to the VoIP driver 220.

<u>FIG. 9 - PSTN TO MOBILE CALL SETUP SEQUENCE</u>

1. PSTN user dials PSTN telephone 340.

2. PBX telephone system 130 receives call setup request via PSTN 160, and routes request to gateway server software 175.

3. Gateway server 110 validates the direct inward dialing (DID) or extension number and finds that this user currently has a mobile phone 285 that is ON. It passes on the call setup request, along with mobile phone ID and other user identification information (such as user's name, for caller ID), to the wireless gatekeeper server 295.

4. Wireless gatekeeper server 295 looks up which wireless base station 290 is currently associated with the mobile phone 285 being called and requests that wireless base station to ring through to the mobile phone.

5. Called mobile user answers. This information is relayed from the wireless base station 290 back to the wireless gatekeeper server 295.

6. The wireless gatekeeper server 295 responds to the gateway server 110 that the call is connecting OK.

7. The gateway server 110 completes the call circuit setup, configuring the PBX telephone system 130 to begin routing voice packets via the trunk driver 240 to the VoIP driver 220.

8. Meanwhile, the wireless base station 290 connects with the VoIP Driver 220.

9. Wireless base station 290 coordinates with mobile phone 285 to start routing IP voice packets to the VoIP driver 220.

FIG. 10 MOBILE-TO-PBX ROUTING WITH SAME GATEWAY NETWORK

In FIG. 10 a mobile phone user uses the mobile phone 285 to call another employee on the campus (or in same building). The called user is on the PBX telephone system 130. Note the caller's voice is transmitted to the wireless base station 290 where is it digitized, compressed, packetized into H.323 data packets which are then deposited on the LAN 140 and routed to the gateway server's VoIP driver 220. The VoIP driver 220 and its' associated digital signal processor (DSP) decompress the IP packets' voice data and push the data across a time division multiplexed (TDM) bus 355 where is it sent to the PBX telephone system 130 via an analog or T1/E1 digital trunk 360 coupled to the PBX 250. From the PBX telephone system 130 the analog voice goes to the PBX telephone 300. Note, that for purposes of clarity, details of the TDM bus 355 and the T1/E1 trunk 360 that are widely known have been omitted from the figures and the description. However, TDM buses and T1/E1 trunks and how they work are discussed in greater detail in U.S. Pat. App. Serial No. 09/061,802.

The reverse of this process flow occurs from the PBX telephone 345 back to the mobile phone 285 as well. Note that the same codec algorithm must be used in both the gateway server's VoIP driver 220 and in the wireless base station 290.

If the caller is traveling with his/her mobile phone 285, a handoff may be required if the user leaves the range of one wireless base station 290 and enters the range of another. This will be handled via H.323 messaging between the wireless base stations 290 and the wireless gatekeeper server 295; the wireless gatekeeper server must inform the gateway server 110 of the IP address of the new wireless base station so voice data can be properly routed.

In another aspect, Quality of Service (QoS) can be monitored for the LAN and the VoIP packets, and an alert is provided that could be forwarded to a network management system 350, where a suitable action might be taken to correct the problem. However, an occasional QoS problem resulting from network congestion is not likely to be an issue, given the expectations of wireless users.

FIG. 11 - MOBILE-TO-PBX CALL SETUP WITH SAME GATEWAY NETWORK

In order to set up the call, the caller's mobile phone 285 first performs a handshake between itself and the wireless base station 290. The exact format and data exchanged in this handshake varies according to the wireless protocol being used, but generally involves the mobile phone 285 transmitting its mobile ID (usually a ten digit telephone number) and other identifying information. The wireless base station 290 then converts the identification information to H.323 signaling format and sends it to the wireless gatekeeper server 295, which validates that the mobile phone 285 is allowed access to the private-wireless-network 120. The wireless gatekeeper server 295 routes the access request to the gateway server software 175, along with the IP address of the wireless base station 290. The gateway server 110 may also fetch additional information about the handling of the call. For example, it can update its internal tables and call log, and has the PBX telephone system 130 set up the call to the requested extension.

FIG. 12 - MOBILE-TO-MOBILE ROUTING WITH SAME GATEWAY NETWORK

FIG. 12 shows a first mobile phone 285A being used to call another employee on the same campus (or in same building) who is also using a second mobile phone 285B. Note the caller's voice is transmitted to the wireless base station 290 where is it digitized, compressed, packetized into H.323 data packets which are then deposited on the LAN 140 and routed directly to another wireless base station 300. The called party's wireless base station 300 then de-packetizes the IP voice packets, decompresses the data, and converts it back to the format compatible with the called mobile phone 285B, and transmits it out to the called mobile phone. Of course, the same flow occurs in the reverse direction as well.

FIG. 13 MOBILE-TO-MOBILE SETUP WITH SAME GATEWAY NETWORK

When the caller dials using, for example, the office extension for the called party, and then pushes a mobile phone's "SND" button, the wireless base station 290B currently associated with the called mobile phone 285B is notified of the call setup request. This request is forwarded to the wireless gatekeeper server 295, which may authenticate the mobile identification of the caller mobile phone 285A. Assuming the authentication passes, the wireless gatekeeper sever 295 forwards the call setup request with the caller

mobile identification to the gateway server software 175. The gateway server software 175 checks its database 180 to obtain the user information for the caller, and it also uses the dialed office extension to look up the current location of the called party and checks whether the called party has any special call forwarding enabled. The gateway server 110 determines that the user's mobile phone 285B is currently operating, obtains the corresponding mobile identification of the called party, and routes the call setup request with mobile identification of the called party back to the wireless gatekeeper server 295. The wireless gatekeeper server 295 then determines which wireless base station 290B the mobile phone 285B is currently associated with, and routes the call setup request to that wireless base station. The called wireless base station 290B dials the called party's mobile phone 285B. The called wireless base station 290B now begins routing IP packets to the caller wireless base station 290A (which was known to it via the call setup request). The success of the call setup operation is communicated from the called wireless base station 290B back to the wireless gatekeeper 295, which now informs the caller wireless base station 290A of the address of the called wireless base station, and the caller wireless base station can now begin routing its voice IP packets.

FIG. 14 - MOBILE-TO-PSTN ROUTING DIFFERENT GATEWAY NETWORKS

FIG. 14 shows a scenario for routing a telephone call over the IP Network 145 which will typically allow the users to bypass toll charges that would be incurred using the PSTN or a public-wireless-network.

In FIG. 14 a first user or employee uses a mobile phone 285 to call another employee located at another, remote company site, and wherein the called employee is on a PSTN telephone 340 that is located near the remote company site. The caller's voice is transmitted to the wireless base station 290 where is it encoded and packetized into H.323 IP data packets which are then deposited on the LAN 140 and routed to the IP network 145. At the remote site, the called gateway server's VoIP driver 220 de-packetizes and decodes the IP packets' voice data and sends the data across the TDM bus to the trunk driver 225 where is it sent to the PBX 250 via an analog or digital T1/E1 trunk 360. From the PBX 250 the voice is routed to a first telephone company central office (CO1) 370 in the PSTN. CO1 370 will route the data to another telephone

company central office (CO2) 375 that is serving the called user's telephone. Finally, the voice data reaches the called party using the PSTN telephone 340.

The reverse of this process flow occurs from the PSTN telephone 340 back to the mobile phone 285 as well, so the called user's voice can flow back to the caller. Note that the same codec algorithm must be used in both the gateway server's VoIP driver 220 and in the called user's private-wireless-nctwork 120.

Note that a mobile phone user in the private-wireless-network 120 can also use this scenario to directly dial a PSTN telephone number, where the gateway server 110 near the caller identifies (via the dialed telephone number) which other gateway network 105 is closest in location to the targeted PSTN telephone. In either case, the routing allows the bypass of toll charges when the called party is located in the PSTN near a remote gateway network 105; this type of call routing is termed a "hop-off to PSTN".

FIG. 15 - MOBILE-TO-PBX ROUTING DIFFERENT GATEWAY NETWORKS

FIG. 15 shows a second scenario for routing a call in which the routing of the call over the IP Network 145 will typically allow the users to bypass toll charges.

In this figure a mobile phone user uses the mobile phone 285 to call another employee located at another company site, and the called employee is on a PBX telephone 345. The caller's voice is transmitted to the wireless base station 290 where is it encoded and packetized into H.323 IP data packets which are then deposited on the LAN 140 and routed to the IP network 145. At the remote site, the gateway server's VoIP driver 220 de-packetizes and decompresses the IP packets' voice data and sends it to the PBX 250 via the TDM bus 355, trunk driver 240 and an analog or digital T1/E1 trunk 360. From the PBX 250 an analog voice signal is transmitted to called user's PBX telephone 345.

The reverse of this process flow occurs from the PBX telephone back to the mobile as well. Note that the same codec algorithm must be used in both the gateway server's VoIP driver and the wireless base station.

FIG. 16 - MOBILE-TO-PBX SETUP DIFFERENT GATEWAY NETWORKS

FIG. 16 shows the call setup steps that might result in the call routing shown in FIG. 15. In this case the mobile caller is homed in the private-wireless-network 120 from which the caller is originating the call. The call is originated at the caller's mobile phone

285 which coordinates with the wireless base station 290. The wireless base station 290 forwards the call setup request to the wireless gatekeeper server 295, which may verify that the mobile phone 285 is allowed access to the private-wireless-network 120. The wireless gatekeeper server 295 in turn routes the call setup request to the gateway server software 175A. The caller gateway server software 175A inspects the dialed digits and determines that the called party is homed at another gateway network 105B. The caller gateway server software 175A then queries the gateway server 110B in the called gateway network 105B for the routing information of how to get to the called telephone. In this case, since the called party is at a PBX telephone 345 of the called gateway server 110B, the called gateway server responds with the IP routing address of the VoIP driver 220B of the caller gateway server. The caller gateway server software 175A then relays this information back to the wireless gatekeeper server 295, which then coordinates with the wireless base station 290 to set the routing of voice IP packets directly to the called gateway server's VoIP driver 220B. The called VoIP driver 220B routes the call setup request to the PBX 250 via the station and trunk drivers 240, 225, and finally on to the called party's PBX telephone 345.

FIG. 17 - MOBILE-TO-MOBILE ROUTING DIFFERENT GATEWAY NETWORKS

FIG. 17 shows a third scenario for routing a call in which the routing of the call over the IP Network 145 will typically allow the users to bypass toll charges.

The-voice IP packets are generated at the caller wireless base station 290A, and routed directly to the called wireless base station 290B via the IP network 145. The called wireless base 145B station de-packetizes the voice IP packets and routes the data to the called mobile phone 285B.

FIG. 18 - MOBILE PHONE BEING TURNED ON IN Private-wireless-network

When the mobile phone 285 is turned on when the telephone's user is within the coverage area of the private-wireless-network 120 the mobile performs a registration process with the nearest wireless base station 290. The wireless base station 290, if it hasn't seen this user in at least some time period (a period similar to the registration interval of the mobile phones, for example 10 minutes), will send a message to the wireless gatekeeper server 295 to inform it of the mobile phone's presence. Once the

wireless gatekeeper server 295 authenticates the ID of the mobile phone 285, it sends on a notification to the gateway server 110 that this particular mobile is now available for calls. The gateway server 110 checks its database to get the user and location information associated with the mobile phone 285. If the particular mobile phone 285 is the secondary or backup telephone of an on-site employee's office telephone, the gateway server 110 will update its database with this information, so that any subsequent calls to the employee's office extension can be automatically forwarded by the gateway server to the mobile phone. If the mobile phone 285 is an employee's primary (only) telephone, the status of the user is updated so that, for example, the user can now be called at the mobile phone instead of routed off to voicemail. If the gateway server 110 determines that the mobile phone 285 is associated with a visiting user from a different company site, the gateway server notifies the remote gateway server at the visitor's home location regarding the new location of its mobile phone; the remote gateway server stores this information for the purpose of routing subsequent calls for the user.

It should be noted that this scenario of a mobile phone 285 being turned on is the same scenario as occurs if a user comes back in range of the gateway network 105 after having been out of range, and with the assumption that there is no roaming between the private and public-wireless-networks 120, 150. Such a scenario might occur, for example, if an employee goes out to lunch and forgets to turn off the mobile phone which he carries with him.

<u>MOBILE PHONE BEING TURNED OFF IN Private-wireless-network</u>

The sequence of notification events for a mobile phone 285 being turned off is virtually identical to the sequence of events for a mobile phone being turned on, which is illustrated in FIG. 18.

When the wireless base station 290 to which a mobile phone 285 has been registering determines that the mobile phone has been turned off, or has not registered or made calls for some time period (generally a period greater than the registration interval of the mobile phone), then this information is forwarded to the wireless gatekeeper server 295. The wireless gatekeeper server 295 may verify that the mobile phone 285 has not appeared at another wireless base station 290, and forwards the notification that the mobile phone is no longer present to the gateway server 110. As with the case of a

mobile phone 285 being turned on, the local database 180 is updated to note that the mobile phone is no longer available for calls. If the mobile phone 285 that went off the air belongs to a visitor from another site, then the notification is also passed to the remote gateway server at the visitor's home site.

It should be noted that this scenario of a mobile phone 285 being turned off is the same scenario as occurs if a user exits from the range of the gateway network 105, and with the assumption that there is no roaming between the private and public-wireless-networks 120, 150. This might occur, for example, if an employee goes out to lunch and forgets to turn off the mobile phone 285 which he carries with him.

FIG. 19 - MOBILE PHONE TURNED ON IN Public-wireless-network, MOBILE HOMED AT LOCAL SITE

It is important to the proper operation of the gateway server 110 in conjunction with the private-wireless-network 120 that the gateway server maintain the knowledge of (1) where (i.e. at which site or associated gateway network 105) all the operating mobile phones 285 are homed, and (2) which mobile phones (either homed locally or roaming from another gateway network) are currently operating in the gateway network's wireless private network 120, and (3) which mobile phones (either homed locally or roaming from another gateway network) are currently operating in the gateway network's nearby wireless public network 150.

The scenario for a mobile phone 285 being turned on in the wireless public network 150, where the mobile phone's home site is in the locale of the gateway network 105 is shown in FIG. 19. Referring to FIG. 19, the mobile phone 285 is turned on and registered in the public-wireless-network 150. The public-wireless-network 150 determines this is a roamer mobile phone 285 and routes the information of the presence of the mobile phone to the wireless gatekeeper server 295 in the private-wireless-network 120 via a signaling link 380 connecting the two wireless-networks. The wireless gatekeeper server 295 then routes the notification to the gateway server software 175, so it can update its database 180 and properly route subsequent calls to the mobile phone 285.

FIG. 20 - MOBILE PHONE TURNED ON IN Public-wireless-network, MOBILE HOMED AT REMOTE SITE

The scenario for a mobile phone 285 roaming in the locale of another gateway network 105 is slightly different. The mobile phone registration is received in the public-wireless-network 150. Based on identifying information which is part of the registration data, the public-wireless-network 150 knows to route thc registration notification to the wireless gatekeeper sever 295 in the mobile phone's home private-wireless-network 120. The wireless gatekeeper sever 295 notifies the gateway server software 175 in the associated gateway server 110 of the mobile phone's registration. The gateway server software 175 updates its database 180 with the new location information of the mobile phone, and, because it is further determined that this mobile phone is operating from a remote location, any other gateway network 105 associated with that location is also notified that this mobile phone is operating in its area.

FIG. 21 - MOBILE IN Public-wireless-network CALLING TO LOCAL PBX TELEPHONE

FIG. 21 shows the routing that results from a mobile phone 285 in the public-wireless-network 150 calling to a PBX telephone 345, where both telephones are associated with the same gateway network 105. The call path starts at the mobile phone 285, is routed through the public-wireless-network 150 and the PSTN 160 to the PBX 250, where it is routed directly to the PBX telephone 345.

FIG. 22 - CALL SETUP FROM MOBILE PHONE IN Public-wireless-network TO PBX TELEPHONE

When a mobile phone 285 dials to a company internal office PBX telephone 345, using DID for example, the public-wireless-network 150 receives the call setup request, forwards it into the PSTN 160, which in turn dials into the PBX 250. The PBX 250 sends the call setup request on to the gateway server software 175, which looks up the incoming dialed number in its database 180 to determine where the user for this number is currently located, or whether and where the user has requested the call be forwarded to. Once the software 175 determines that the called party should be dialed at a local PBX telephone

345, it responds back to the PBX 250, which completes the call routing through to the PBX telephone.

FIG. 23 - PBX TELEPHONE CALLING TO MOBILE PHONE IN Public-wireless-network

The call routing for this scenario is similar to the call routing described for the that shown in FIG. 21 except that two channels on the PBX 250 are used. This is due to the fact that once a telephone call is routed to the gateway server 110, the PBX 250 does not support re-routing of that leg of the call that has already been set up to the gateway server.

FIG. 24 - CALL SETUP FROM PBX TELEPHONE TO MOBILE PHONE IN PUBLIC TELEPHONE NETWORK, MOBILE HOMED LOCALLY

The caller at the PBX telephone 345 initiates the call to another employee homed at the same office, typically by dialing the office extension of the called party. The call setup request is routed from the PBX telephone 345 to the gateway server software 175 via the PBX 250. The gateway server software 175 looks up the called user in its database 180 and finds that the user with this extension currently has an active mobile phone 285, and that this telephone is currently operating in the public-wireless-network 150. Because the public-wireless-network 150 may be re-assigning the mobile phone temporary mobile identification numbers at regular intervals, the gateway server now queries the wireless gatekeeper server 295 for the current temporary mobile identification of the mobile phone. The wireless gatekeeper server 295 maintains this information via its signaling link 280 to the public-wireless-network 150. The wireless gatekeeper server 295 then returns the requested mobile identification information to the gateway server software 175 which has the PBX 250 dial out to the PSTN 160 using the mobile identification obtained from the wireless gatekeeper server.

FIG. 25 - CALL SETUP FROM PBX TELEPHONE TO MOBILE PHONE IN PUBLIC TELEPHONE NETWORK, MOBILE HOMED REMOTELY

In this scenario, the caller at the PBX telephone 345 initiates the call to another employee homed at a different office by dialing the location ID and office extension of the called party. The call setup request is routed from the PBX telephone 345 to the

gateway server software 175 via the PBX 250. The gateway server software 175 looks up the called user in its database 180 and finds that the user with this extension is homed at a remote site and currently has an active mobile phone 285, and that this mobile phon is currently operating in the public-wireless-network 150. Because the public-wireless-network 150 may be reassigning the mobile phone 285 temporary mobile identification numbers at regular intervals, the gateway server 110 now must obtain this temporary mobile identification. It does so by querying the mobile phone's home gateway server 110 via the IP network 145. The remote gateway server 110 in turn queries its wireless gatekeeper server 295 for the current temporary mobile identification of the mobile phone 285. The wireless gatekeeper server 295 maintains this information via its signaling link 380 to the public-wireless-network 150. The wireless gatekeeper server 295 then returns the requested mobile identification information to the remote gateway server software 175, which is then returned to the original requesting gateway server software 175, which in turn has the PBX 250 dial out to the PSTN 160 using the mobile identification obtained from the wireless gatekeeper server.

FIG. 26 - PBX TELEPHONE CALLING TO PRIVATE MOBILE IN Public-wireless-network AT A REMOTE SITE

In the routing shown in FIG. 26, the caller at a PBX telephone 345 at one site is calling an employee at another site who is using his mobile phone 285 the public-wireless-network 150. The call is routed through the PBX 250 to the gateway server 110 via the trunk driver 225. The trunk driver 225 routes the call data across the TDM bus 355 to the VoIP driver 220. The VoIP driver 220 converts the data into IP packets and routes it across the IP network 145 to the remote gateway server's VoIP driver 220. The remote VoIP driver 220 reverses the process of the other VoIP driver, extracting the data from the IP packets and routing the voice to the trunk driver 225, which in turn routes it to the PBX 250 and then out to the PSTN 160. The PSTN 160 is connected with the public-wireless-network 150 which in turn routes the call through to the called mobile phone 145.

FEATURES

The communication system 100 of the present invention is adapted to provide a

plurality of features among the several gateway networks 105 at different sites over the IP network 145 regardless of the configuration of the gateway networks or the telephones used. These features can bc broadly classified as automated call control features and as usability features. The automated call control features include quality of service monitoring, fallback to the PSTN 160, monitoring of mobile phones 285 and support of multiple different private-wireless-networks 120. The usability features include support for multiple different numbering plans, caller ID, call control features available at mobile phones 285, call control at browser applications, support for handling a busy destination at browser application, support for a phone directory at a browser application, a call log, follow me control and integrated operations, administration, and maintenance, all of which are described in greater detail below.

Automated Call Control Features

Quality of Service Monitoring

As described in U.S. Pat. App. Serial No. 09/061,802, which is incorporated herein by reference, there are several quality of service indicators which can be placed in the IP voice packets at the time when the packets are generated in preparation for routing onto the LAN 140. These quality of service indicators include indicators to test for delay in transmission of data packets, losses in transmission of data packets, or jitter. At the later point in the call path routing, when the data in the IP packets are extracted and decoded, the quality of service indicators are assessed to determine if a reasonable quality for data or voice conversation is being maintained. In the VoIP call routing paths discussed in the above referenced application, the voice IP packets are always generated/encoded by the VoIP driver in one gateway server and extracted/decoded by a VoIP driver in another gateway server.

In the communication system 100 of the present invention, with the introduction of the private-wireless-network 120, the quality of service (QoS) determination may involve two separate legs of a call routing path over the IP network 145. For example, consider the call path shown in FIG. 15; the voice IP packets that are generated at the wireless base station 290 are de-packetized and decoded at the VoIP driver 220. Thus, it is important that the multiple vendors agree on which QoS indicators to use. Assuming that both the VoIP driver vendor and the base station vendor agree on how to use QoS,

then we need to assess where in the call path the quality degradation is taking place. In many cases, such as that shown in FIG. 15, the degradation may occur on the IP Network 145, or it may occur on the LAN 140, along the routing leg between the wireless base station 290 and the LAN. Since there is a method to improve degraded performance of the IP network, it is most useful to detect a IP Network problem.

In order to detect a IP Network problem, the VoIP driver 220 of the invention continues to monitor QoS for the entire path segment over which voice IP packets travel. Once the QoS for this entire segment degrades to a given threshold, a secondary simple test of the QoS between the two gateway servers 110 (the gateway server of the caller and the gateway server of the called party) will be executed. This secondary test may involve sending bursts of IP packets over the IP network 145 and having the VoIP drivers 220 on either end assess QoS. Alternatively, the test may be more simplistic, such as issuing a series of "ping" commands across the IP network. If it is found that the secondary QoS is also degraded past a threshold, then the fallback to PSTN process can be initiated. If it is found that the secondary QoS indicates no degradation, then the LAN 140 is having the problem. In this case there is little that the gateway server 110 can do to alleviate the problem, other than alarm to an external system, such as a network management system (not shown). Such an alarm, in more sophisticated network configurations, could activate load balancing algorithms which might eliminate high-load network activities or users.

Fallback to PSTN

Numerous different methods for providing fallback to PSTN in the event of diminished QoS are described in U.S. Pat. App. Serial No. 09/061,802, which is incorporated herein by reference. The differences between the methods relate primarily to the variations in PBX configurations. The methods in the above cited reference are generally the same when applied to the PBX telephone 345 at one end of the call. However, for scenarios having a mobile phone 285 at the one end of the call, some additional steps must be taken. A method for enacting fallback in a VoIP call routing path involving a mobile phone 285 will now be described with reference FIG. 15.

Consider that FIG. 15 illustrates an original VoIP call involving a mobile phone 285, and that FIG. 27 illustrates the modified call path once the fallback to PSTN operation has completed. Referring to FIG. 15, a fallback to PSTN 160 is accomplished

as follows. The sequence of events starts when the VoIP driver 220 in called gateway server 110B detects QoS degradation. It performs a secondary test by exchanging packets between its VoIP driver 220 and the VoIP driver in the caller gateway server 110A. If this test also indicates degradation, the called gateway server 110B initiates a fallback to PSTN 160. First, the called gateway server 110B announces its intention via a notification sent to the caller gateway server 110A. The caller gateway server 110A then coordinates with the called gateway server 110B to set up a PSTN call between the two gateway servers via the PBXs 250A, 250B. As soon as the PSTN call is set up between the two gateway servers 110A, 110B, the caller gateway server must command the caller wireless base station 290 to start routing its packets to the local VoIP driver 220A instead of the remote one.

Gateway Server Monitoring of Operating Mobiles

In one embodiment of the invention, the gateway server 110 must retain the knowledge of which mobile phones 285 are operating (e.g., turned on) in the private-wireless-network 120. The wireless gatekeeper server 295 must keep the gateway server 110 apprized of the status of the mobile phones 285 in several categories including: (1) mobile phones in the home private-wireless-network 120 (2); mobile phones in the local public-wireless-network 150; (3) roamer mobile phones in the home private-wireless-network; and (4) roamer mobile phones in the local public-wireless-network.

Monitoring of Mobiles in the Public Network

The monitoring of mobile phones 285 in the public network 150 is more involved, since these mobile phones may be given temporary mobile ID's in some public-wireless-networks. When updating the gateway server 110 regarding mobile phones 285 on the public network, the wireless gatekeeper server 295 must provide not the temporary mobile ID, but rather the mobile ID by which the mobile is known when operating within the private-wireless-network 150. This allows the gateway server 110 to determine which mobile phones 285 are active, for subsequent routing purposes.

In other wireless-network arrangements, the mobile ID of the mobile phones 285 may be unique within both public and private-wireless-networks 150, 120.

Multiple Private-wireless-nctwork Vendors Supported

In another aspect, the communication system 100 of the invention is capable of supporting multiple privatc-wireless-networks 120 manufactured by different vendors. For example, the wireless-network equipment associated with one gateway server 110 may differ from the wireless equipment vendor associated with another gateway server. For this to be accomplished, both private-wireless-networks must support and be using the same kind of encoding and compression algorithms (such as enhanced full rate GSM codec), and they further must support the same voice over IP protocol, such as H.323.

Usability Features

Support for Multiple Different Numbering Plans

Numerous different methods for providing support for multiple different numbering plans are described in U.S. Pat. App. Serial No. 09/061,802, which is incorporated herein by reference. The difference between the methods in the above cited reference, and methods for a communication system 100 having a mobile phone 285 within a private-wireless-network 120, is that the private-wireless-network itself needs to maintain knowledge of the mobile phones it serves in terms of the telephone's mobile identification number.

Generally, the mobile identification number is a 10-digit string which uniquely identifies the mobile phone 285 within the private-wireless-network 120 and within all roaming partners of the wireless-network. When the mobile phone 285 is used as an employee resource within a company, it is most convenient for users not to have to remember a separate telephone number for the mobile phonc, which a caller might dial after dialing to the office extension fails because the employee didn't answer. In the present invention, the employees and the users of the mobile phones 285 are not required to remember mobile phone numbers. Instead the gateway server 110 automatically maps a dialed number to the mobile phone number, and the mobile phone number need only be known within the system 100.

For example, if an employee caller dials another employee using an ETN number, i.e., 8+Location+Extension, the gateway server software 175 receives the call setup request and looks up which user has the given Location + Extension. If the called user does not have an actual PBX telephone 345 but does have an operating mobile phone 285,

then the 10-digit mobile identification for the user's mobile phone is automatically passed to the relevant wireless gatekeeper server 295 along with a request to set up the call. If instead the mobile phone 285 (of the called user who doesn't have a PBX telephone 345) is not currently operating, then the gateway server 110 can route the call directly to the called party's voicemail. If the called user has both a PBX telephone 345 and a mobile phone 285, the gateway server 110 configuration (or the individual user's configuration) may specify that if the mobile phone is operating, a call attempt to the mobile phone be initiated; and if the mobile phone is not operating, the call be attempted to the PBX telephone.

Since the mapping of the dialed number to the ten-digit mobile identification number is handled entirely within the gateway server 110, the dialing plan for the mobile phone 285 can be configured for any of the plans discussed in the original patent. Furthermore, the convenience of having to recall only a single telephone number for each user is a unique productivity enhancement offered by the invention as it encompasses a wireless capability.

A configuration where some users in a network are equipped with IP telephones 320 as their primary telephones, while others have PBX telephones 345, can also be handled seamlessly by the present invention. The dialing plan can be coordinated such that both kinds of telephone users have the same location ID, with possibly different extension ranges. Users at other sites make calls to both kinds of telephones, dialing 8 + location B) + extension, and the callers don't need to know which kind of telephone a particular user has.

Caller ID

As described in U.S. Pat. App. Serial No. 09/061,802, the integration of a gateway server 110 with an enterprise directory 205 provides a central repository for modifying user information. User information includes many different attributes describing the particular user, including name, and office extension number for the user, etcetera. In the present invention, additional attributes are added to support the additional user information required to be maintained regarding a user's mobile phone 285 and a user's IP telephone 320. Table II shows a set of additional attributes that are used to support the wireless private network's inclusion in the gateway network 105.

TABLE II

Attribute	Description
Primary telephone Type	One of"PBX,', "Mobile", or "JP"
Office Mobile Number	10-digit mobile identification number
Office Mobile Equip ID	Equipment ID that uniquely identifies the telephone; for GSM telephones, this is the IMEI (International Mobile Equipment Identity); for AMPS telephones, this is the ESN (Electronic Serial Number).
Authentication Key	Key used for verifying the identity of the mobile phone user.
Private Roaming Enabled	True/False indicator of whether the mobile phone is allowed to be used outside its home private-wireless-network, i.e. in private-wireless-networks at other company sites.
Public Roaming Enabled	True/False indicator of whether the mobile phone is allowed to be used in the public-wireless-network.
IP telephone IP Address	IP address of the user's primary IP telephone.

On Mobile phone Incoming Calls

Since all the call setup for calls made to the mobile phones 285 is done under the coordination of the gateway server software 175, which accesses the enterprise directory 205, caller ID information is always added to the call setup request, which flows from the gateway server 110 to the wireless gatekeeper server 295 to a wireless base station 290, where it is finally made available to the mobile phone. Of course, the display of caller ID information is also dependent on what a handset of the mobile phone 285 can support, and whether the wireless protocol being used by the vendor of the wireless-network equipment supports the exchange of caller ID information; for example, GSM Phase 2 supports caller ID, but GSM Phase 1 does not.

On Other Telephones, Outgoing Calls from Mobile

Outgoing calls from the mobile phone 285 can support the caller ID, as the mobile ID is guaranteed to be routed to the gateway server 110, which can then look up the caller information in its database 180 and forward it to the called party, and the information can thus be displayed as described in U.S. Pat. App. Serial No. 09/061,802.

On IP Telephones Incoming Calls

Since all the call setup for all calls made to the IP telephones 320 is done under the coordination of the gateway server software 175, which accesses the enterprise directory 205, caller ID information is always added to the incoming call setup request, which flows from the gateway server 110 to the IP call manager 315 to the IP telephone 320, where it is shown on the IP telephone display (not shown).

On Other Telephones, Outgoing Calls from IP Telephone

Outgoing calls from the IP telephone 320 can support the caller ID, as the call is guaranteed to be routed to the gateway server 110, which can then look up the caller ID information in its database 180 and forward it to the called party, and the information can thus be displayed as described in the original invention.

At Browser Application, Incoming Calls

Calls to either IP telephones 320 or mobile phones 285 are routed through the gateway server 110. The gateway server 110 has the knowledge as to which users currently are logged into the gateway server via a browser GUI interface 385 on the users' PCs. If the browser GUI interface 385 is up for the given user, the gateway server 110 will cause the browser GUI interface to display the caller ID information for the call as part of a pop-up dialog window (not shown).

Call Control Features Available at Mobile Phone

Since the mobile phone 285 is, in many instances, a convenient alternative to a PBX telephone 345, in one aspect the communication system of the present invention is adapted to make the same call control features generally available to the user on PBX telephone are also be available to the user on the mobile phone. These features include,

call hold/retrieve, transfer, forwarding and conferencing, all of which are described in more detail below.

Hold/Retrieve

If the private wireless system 120 supports it, a second call can be routed to the user on a mobile phone 285 while a call is already in progress. The second call may be signaled to the mobile phone user by a special tone played over the incoming voice, and the caller ID for the new caller may be shown on the display of the mobile phone 285. If the mobile user wishes to, he may verbally ask the original caller to hang on, and then dial a sequence, such as "*5", to put the first caller on hold and answer the new caller. Subsequently, the mobile phone user may dial another sequence, such as "*6" to retrieve the original and resume that conversation.

Transfer

The transfer of a call from the called party using a mobile phone 285 to another caller in the enterprise can be accomplished by the invention under coordination of the gateway server 110. Consider FIGs. 28 and 29. FIG. 28 shows a called in progress from a caller on a PBX telephone 345 to a called party using a mobile phone 285 in the private-wireless-network 120 of the same gateway server 110. FIG. 29 shows the end result of the mobile phone user's action to transfer the call to another user within the enterprise. In this case the second called user is located at another site, and thus the resulting routing is accomplished via the IP network 145.

Exemplary steps describing how the transfer of the call is accomplished by the communication system 100 are shown in FIG. 30 and described in the following sequence of steps:

1. In conversation with the caller, called party 1 (in FIG. 28) determines that the caller should be transferred to speak with the called party 2 (in FIG. 29). So the caller keys a special sequence, such as "*3" followed by the location ID and extension of the called party 2.

2. The key sequence is interpreted by the wireless base station 290 which forwards the transfer request to the wireless gatekeeper server 295.

3. The wireless gatekeeper server 295 in turn forwards the transfer request to the gateway server software 175.

4. The gateway server software 175 forwards the setup request to the remote gateway server software with the given location ID defined in the dialed digits and the extension.

5. The remote gateway server software 175 checks its database 180 and determines that the called party is in its gateway network 105. The remote gateway server software 175 sets up the routing for the new call via its PBX 250, and dials through to the PBX telephone 345.

6. The remote gateway server 110 commands its VoIP driver 220 to route packets for the call to the other gateway server's VoIP driver.

7. The remote gateway server software 175 sends a response back to the first gateway
server's software with its VoIP address and the information that the other end of the call has been routed.

8. The gateway server software 175 commands its VoIP driver 220 to start routing the voice packets to the VoIP driver in the remote gateway server via the IP network 145.

9. The gateway server software 175 messages to the wireless gatekeeper server 295 that it can take down its leg of the call.

10. The wireless gatekeeper server 295 relays this information to the wireless base station 290.

11. The wireless base station 290 terminates its call with the mobile phone 285.

Forwarding

A method for forwarding a call using the gateway server 110 is described in U.S. Pat. App. Serial No. 09/061,802. The method can be extended to handle communication systems 100 having mobile phones 285. Consider, for example, a user who has set a "follow-me" function to route calls first to his mobile phone, and secondly to a telephone at a secretary's desk. An incoming call is routed by the gateway server 110 to the wireless gatekeeper server 295, and in turn to the wireless base station 290 with which the mobile phone 285 is currently registering. If after several rings the mobile user doesn't answer, the wireless base station 290 routes a negative acknowledgment back to the wireless gatekeeper server 295, which in turn routes it back to the gateway server 110. The

gateway server 110 receives the acknowledgment and then routes the call via the PBX 250 to the secretary's desk telephone.

Conferencing

Call conferencing is also coordinated through the gateway server 110. If both the wireless telephone system 120 and the PBX 250 and associated PBX telephone 345 support conference calling, then conferencing of multiple callers can be supported. FIG. 31 shows an example of a 3-way conference call in progress. The first caller at the PBX telephone 345 has called a co-worker at a mobile phone 285A (the second party). After their call was established, a third party, calling from another mobile phone 285B at a remote office, joins the conversation. In the implementation shown, there are three call paths set up. For example, the second party has two call paths set up to it. In the wireless system shown, the base station is combining the two incoming voice streams as needed, and funneling the combined stream to the mobile phone 295. At the first party's PBX telephone 345, the CTI services module 200 may be working with the PBX 250 to implement the conferencing, for example, by selecting the higher volume of the two incoming voices and patching that voice to the first party. With the gateway server110 of the present invention, the usage protocol for conferencing can be offered in the same way for the different kinds of telephone users.

Call Control At Browser Application

Call control features can also be offered to the mobile phone user who is logged into the LAN 140 and running a browser application that provides a browser GUI interface 385 to the gateway server 110. The mobile phone 285 may be the user's primary telephone, and for convenience, the user may want to control the telephone's features through the browser GUI on the user's personal computer.

FIG. 32 illustrates an example wherein a user clicking on the "dial" button in the PC browser GUI interface 385, after having selected another user to dial from the enterprise directory 205. The PC browser messages to the gateway server software 175 that the user wants to initiate a call (step 1 in the figure), and to whom. The message also contains the identification of the calling user, and so the gateway server 110 checks its database 180 and finds that the primary telephone for the user is the user's mobile phone

285. The gateway server software 175 then messages to the wireless gatekeeper server 295 (step 2), which in turn messages to the relevant wireless base station 290 regarding the call initiation (step 3). The wireless base station 290 coordinates with the mobile phone 285 to have it initiate the call automatically (step 4). From this point forward, the telephone call is treated like any other telephone call. For example, if the telephone call is to a PSTN telephone 340, the call setup would be exactly as described in FIG. 8 above. In FIG. 32, the dotted line represents the call path after it has been set up. After the steps described for that call setup path (in FIG. 8) are completed, the gateway server 110 must update the caller's browser GUI interface 385 to indicate the call has now started (step 5 in FIG. 32).

Most of the other major call control functions are handled similarly, by the gateway server 110 coordinating with the private wireless system 120, and updating the browser GUI interface 385 at the completion of the operation. The call control functions handled in this manner can include dial, answer, hold/retrieve, hang-up, transfer, and conferencing. It is important to note that the private wireless system 120 and the mobile phones 285 must support this functionality in order to provide such call control from the PC. The newer standards, such as the GSM phase 2, are supporting such sophisticated functions, where the older ones did not.

The do not disturb and forwarding functions are handled simply by the browser GUI interface 385 messaging the request to the gateway server 110. Since both these actions are simply state changes that will affect future calls, the gateway server 110 simply notes this information in its database 180 and will use it to make routing decisions when incoming calls arrive for the user.

Destination Busy Handling At Browser Application

Destination Busy handling for mobile phones 285 must be discussed in two different contexts, when the busy destination is a mobile phone, and when the caller initiating the call is on a mobile phone. Both scenarios are supported.

In the case where the originator of the call is on a mobile phone 285 and the caller is using the browser GUI interface 385, and a busy signal is reached at the called party (say the called party is on a PBX telephone 345), the caller is provided with several options by the browser GUI interface including, as described in U.S. Pat. App. Serial No.

09/061,802. Namely, the caller is provided with the options of send alert, request callback, ring through and cancel call, which is equivalent to hanging up.

Referring to FIG. 33, a call flow sequence is shown wherein a first user on a mobile phone 285 with the browser GUI interface 385 running at one location calls a second user at another location, where the second user is on a PBX telephone 345 and also has the browser GUI interface running. The second user is already engaged in a telephone conversation on his PBX telephone 345. The steps are as follows.

1. Mobile phone user dials using location ID plus extension and the wireless base station receives 290 the call setup request.

2. Wireless base station receives 290 routes setup request to the wireless gatekeeper server 295.

3. Wireless gatekeeper server 295 routes the setup request to gateway server software 175.

4. Gateway server software 175 looks up which gateway server 110 serves the dialed location ID and routes the setup request to that gateway server.

5. The remote gateway server software 175 receives the request and already knows that the called party is engaged in a call, so it immediately updates the called party's browser GUI interface with a pop-up message (not shown) regarding who is calling.

6. Assuming, for example, that the user does not respond to the pop-up message, the remote gateway server software 175 notifies the original gateway server software that the called party is busy.

7. The gateway server software 175 routes this information to the browser GUI interface 385 of the caller, and gives the caller several destination busy options, including send alert, request callback, ring through, and cancel call.

8. The caller now enters a one line alert message, such as urgent - client wants to back out of the deal.

9. Gateway server software 175 relays the alert message to the remote gateway server 110.

10. The remote gateway server software 175 displays the alert on the called party's browser GUI interface 385 pop-up window.

11. The called party sees the alert message and responds by quickly putting his other party on hold and answering the mobile phone caller's call by clicking on the answer button in the window.

12. The remote gateway server software 175 requests the call be answered via the CTI driver 210 and the other caller be put on hold.

13. The CTI driver 210 and PBX 250 cooperate to put the other part on hold, and switch the called party's routing in the PBX.

14. The remote gateway server software 175 gives the VoIP driver 220 the new routing information, including the address of the wireless base station 290 where the IP packets are going to be routed.

15. The remote gateway server software 175 acknowledges that all this has been accomplished back to the gateway server software associated with the calling gateway network 105.

16. The gateway server software 175 provides the wireless gatekeeper server 295 with the new address where the IP packets are to be routed (i.e., the remote VoIP driver 220).

17. The wireless gatekeeper server 295 requests the wireless base station 290 to re-route the IP packets to the remote VoIP driver 220.

18. The gateway server software 175 updates the caller's po-up window with the information that the call is now connected. Both parties can hear and speak.

In the event that the called party is speaking on a mobile phone 285 and a new call arrives, the mobile phone user may have the alert displayed on his PC if the browser GUI interface 385 is up and the mobile phone has been marked in the system as the user's primary telephone.

All the destination busy operations can be handled in a manner similar to the setup steps described above.

Phone Directory At Browser Application

For mobile phone users, whose primary telephone is the mobile phone 285, a telephone directory (not shown) associated with the browser can be used for convenience when dialing to other users, or when transferring a call, as described in U.S. Pat. App. Serial No. 09/061,802.

Call Log

Call logs (not shown) and methods of creating the same are described in detail in U.S. Pat. App. Serial No. 09/061,802.

Methods of logging of calls to and from mobile phones 285 and IP telephones 320 in a communication system 100 to create a call log according to the present invention are similar. However, it is critical that the gateway server 110 know when a call is being set up, and when a call is terminated, and also when a call changes state (such as when a call is transferred, for example). By considering the call setup flows for setup of mobile phone calls described above, it is dear that the gateway server 110 has the knowledge of when calls are initially set up, and thus can write the needed log information for the call initiation event. On the other hand, it is not so obvious that the gateway server 110 would be aware of call termination events. For example, consider the case of a mobile-to-mobile call as shown in FIG. 12. A call could be ended entirely within the private-wireless-network 120. In order for the call log to operate seamlessly regardless of the kind of telephone the various users are using, the communication system component that first discovers that the termination has occurred, e.g., the wireless base station 290 in the case of a mobile phone 285 and the IP telephone subsystem 115 software in the case of an IP telephone 320, must send out a notification to its controller, e.g., the wireless gatekeeper server 295 or IP call manager 315, which must in turn route the notification to the gateway server 110.

Follow Me Control

At Browser Application

As described in detail in U.S. Pat. App. Serial No. 09/061,802, there are several options for call control. Those discussed in the above cited reference as relevant to PBX telephones 345 are also relevant to mobile phones 285. In addition, for a communication system 100 according to the present invention having multiple types of telephone systems, part of the data stored for each user will include an identification of which telephone is the primary telephone for the user. The primary telephone might be a PBX telephone 345 in the office (the default), a mobile phone 285 or an IP telephone 320. Which telephone is the primary would be entered into the gateway server database 180 by a system administrator. This knowledge of the user's primary telephone type is used

for "follow-me" routing. The default routing of any incoming call for an enterprise user will always be to route it to the primary telephone; if primary telephone is not answered, the default secondary routing is to voicemail. However, there are several ways that more flexible routing can be enabled.

Firstly, users could be given access to modify their "default" routing via the browser GUI interface 385. For example, if a user's default was (1) PBX telephone 345, and (2) voicemail, the user might modify to (1) PBX telephone 345, (2) mobile phone 285, and (3) voicemail.

Smart Filters for Follow-Me Scheduling

Alternatively, users may be given the ability to set up filters (not shown) on how to route their calls at certain times of the day. For example, the user may want to route telephone calls directly to voicemail during nights and weekend times. This can be accomplished by building call filters. A user can build up a set of favorite call filters and apply these at different times of the day. For example, a software engineer user may want to take only particular calls during the 7 to 9 am time-frame to guarantee uninterrupted work time. A filter such as the following could be constructed by the user, using simple point-and-click methods on the browser-based GUI interface 385:

```
Filter UNINTERRUPTED_WORK_TIME
    If caller is one of the following: ("Spouse, Boss, Child's School)
        then Route to Primary Office telephone, and
        then Voicemail
    Else if caller is (Customer X)
        then Route to "Joe Green"
    Else
        Route to Voicemail
End
```

This filter could then be applied to the hours of 7 to 9 am in the user's calendar.

Filters can also be stacked one upon the other. For example, a user might, in addition to the above filter, have a certain filter applied all the time, such as the following:

```
Filter ALWAYS AVOID
```

If caller is "ex-husband" then

Route to voicemail.

End if

Integration with Calendar Tool

In another embodiment of the invention, the browser GUI interface 385 for filter setup is integrated with an enterprise calendar management system (not shown). Examples of calendar management tools include Novell GroupWise and ON Technology's Meeting Maker. Using the browser GUI interface 385, users can request to have their schedule for a given time period imported from the scheduling tool. Then, using a simple point and click method, users can select a scheduled item (such as a staff meeting) and apply an already set-up filter for the time covered by the meeting.

Integrated OA&M (Operations, Administration, and Maintenance)

Yet another usability feature providing by a communication system 100 having an integrated gateway network according to the present invention is the ability to provide integrated operations, administration and maintenance (OA&M) functions for the communication system. In particular, the invention provides a common browser based interface, by which typical OA&M functions can be managed from a single point. An advantage of this is the consistency of service that can be provided across the multiple different kinds of telephone systems.

User Management

User management is one of the most common administrative functions which is typically carried out on a daily basis at an enterprise. The use of the enterprise directory 205, which is the central point for managing all information about a user or employee, and is used to enter all kinds of information regarding usage of different kinds of telephone systems. For example, an administrator could, from the browser GUI interface 385 to add a new employee/user, including name, department info, e-mail address, and the information that an IP telephone 320 is the user's primary telephone and secondary telephone is a mobile phone 285, the mobile phone's ID, and what location and extension is associated with the user. Previous communication systems 100 have not been

integrated in this way. For example, an administrator might have to log into several different systems, including a call manager server (not shown) for specifying the IP address, another OA&M system (not shown) for setting up the mobile phone 285, and so on.

Alarming and Alarm Monitoring

In yet another aspect, the invention supports open standards for alarms, including SNMP. Alarms from the gateway network 105 can be sent to a commercially available network management system (not shown) , or alternatively all alarms can be monitored on a browser-based OA&M monitor (not shown).

It is to be understood that even though numerous characteristics and advantages of certain embodiments of the present invention have been set forth in the foregoing description, together with details of the structure and function of various embodiments of the invention, this disclosure is illustrative only, and changes may be made in detail, especially in matters of structure and arrangement of parts within the principles of the present invention to the full extent indicated by the broad general meaning of the terms in which the appended claims are expressed.

What is claimed is:

1. A communication system (100) for providing communication between a plurality of sites within an enterprise, the communication system (100) comprising:

a public switched telephone network (PSTN 160);

an internet protocol (IP) network (145);

a public-wireless-network (150); and

a plurality of gateway networks (105) coupled to the PSTN (160), IP network (145) and the public-wireless-network (150) to route a telephone call between a calling and a called party thereover, each of the plurality of gateway networks (105) configured to automatically select over which of the IP network (145), PSTN (160) or the public-wireless-network (150) to route the telephone call.

2. A communication system(100) according to claim 1 wherein at least one of the plurality of gateway networks(105) comprises a private-wireless-network (120), and wherein the gateway network (105) is configured to route a telephone call between the calling and the called party over the private-wireless-network (120) and at least one of the IP network (145) (), the PSTN (160) or the public-wireless-network (150).

3. A communication system (100) according to claim 1 wherein at least one of the plurality of gateway networks (105) comprises an UN-PBX system (135) having a telephone coupled thereto, and wherein the gateway network (105) is configured to route a telephone call between the calling and the called party over the UN-PBX system (135) and one or more of the IP network (145), the PSTN (160) or the public-wireless-network (150).

4. A communication system (100) according to claim 1 wherein each of the plurality of gateway networks (105) are configured to automatically reroute an in-progress telephone call routed over a first call path over the IP network (145) to a second call path over the PSTN (160) if a delay in transmission of data packets, losses in transmission of data packets, or jitter exceeds a specified maximum.

5. A communication system (100) according to claim 4 wherein each of the plurality of gateway networks (105) are configured so that the rerouting of the in-

progress telephone call is substantially transparent to the calling party and to the called party.

6. A communication system (100) according to claim 1 wherein when a called party's telephone is unavailable each of the plurality of gateway networks (105) are configured to automatically set up a telephone call between the calling party and the called party as soon as the called party's telephone becomes available.

7. A communication system (100) according to claim 1 wherein the at least one of the plurality of gateway networks (105) is coupled to a directory server (205) comprising information on parties using the communication system (100), and wherein the gateway network is configured to (i) provide a single-point-of-entry for modifications to the information and (ii) to provide replication of these changes across all enterprise sites.

8. A communication system (100) according to claim 1 wherein each of the plurality of gateway networks (105) is configured to provide information identifying the calling party, and wherein the identifying information can be displayed on a display of a called party's telephone or on a computer screen co-located with the called party's telephone.

9. A communication system (100) according to claim 1 wherein each of the plurality of gateway networks (105) is configured to create a log of incoming telephone calls, call attempts and outgoing telephone calls whether the calling and called parties are using a PBX telephone (345), a PSTN telephone (340), an IP telephone (320) or a mobile phone (285).

10. A communication system (100) according to claim 1 wherein when a called party's telephone is unavailable, each of the plurality of gateway networks (105) is configured to enable the calling party to send a computer message that will be immediately displayed on a computer screen co-located with the called party's telephone.

11. A communication system (100) according to claim 1 wherein when the called party does not answer an incoming telephone call, each of the plurality of gateway networks (105) is configured to enable the calling party to forward the telephone call.

12. A communication system (100) according to claim 1 wherein each of the plurality of gateway networks (105) is configured to enable a user of the system to forward telephone calls to a different telephone according to a time schedule predetermined by the user.

13. A method of operating a communication system (100) to connect a telephone call between a calling party on a first telephone and a called party on a second telephone, the method comprising steps of:

coupling a plurality of gateway networks (105) to one another via a public switched telephone network (PSTN (160)), an internet protocol (IP) network (145) and a wireless-network;

automatically selecting over which of the IP network (145), PSTN (160) or the wireless-network to setup the telephone call; and

routing the telephone call over at least one of the IP network (145), PSTN (160) or the wireless-network to connect the telephone call between the calling party on the first telephone and the called party on the second telephone.

14. A method according to claim 13 wherein at least one of the plurality of gateway networks (105) comprises a private-wireless-network (120), and wherein the step of routing the telephone call between the calling and the called party comprises the step of routing the telephone call over the private-wireless-network (120) and at least one of the IP network (145), the PSTN (160) or the wireless-network.

15. A method according to claim 14 wherein at least one of the first and second telephones is a mobile phone coupled to the private-wireless-network (120) , and wherein the step of automatically selecting over which of the IP network (145), PSTN (160) or the wireless-network to setup the telephone call includes the step of determining whether the mobile phone is turned on.

16. A method according to claim 14 wherein the wireless-network is a public-wireless-network (150) and at least one of the first and second telephones is a mobile phone coupled to the public-wireless-network (150), and wherein the step of automatically selecting over which of the IP network (145), PSTN (160) or the wireless-network to setup the telephone call includes the step of learning whether the mobile phone in public-wireless-network (150) is turned on.

17. A method according to claim 16 further comprising the step of automatically rerouting an in-progress telephone call routed over a first call path over the private-wireless-network (120) to a second call path over the public-wireless-network (150) when the mobile phone needs to roam off the private-wireless-network (120) to the public-wireless-network (150).

18. A method according to claim 16 further comprising the step of automatically rerouting an in-progress telephone call routed over a first call path over the public-wireless-network (150) to a second call path over the private-wireless-network (120) when the mobile phone needs to roam off the public-wireless-network (150) to the private-wireless-network (120).

19. A method according to claim 13 wherein at least one of the plurality of gateway networks (105) comprises an UN-PBX system having a telephone coupled thereto, and wherein the step of routing the telephone call between the calling and the called party comprises the step of routing the telephone call over the UN-PBX system and at least one of the IP network (145), the PSTN (160) or the wireless-network.

20. A method according to claim 13 further comprising the step of automatically rerouting an in-progress telephone call routed over a first call path over the IP network (145) to a second call path over the PSTN (160) if a delay in transmission of data packets, losses in transmission of data packets, or jitter exceeds a specified maximum.

BNSDOCID: <WO 0069156A1_I >

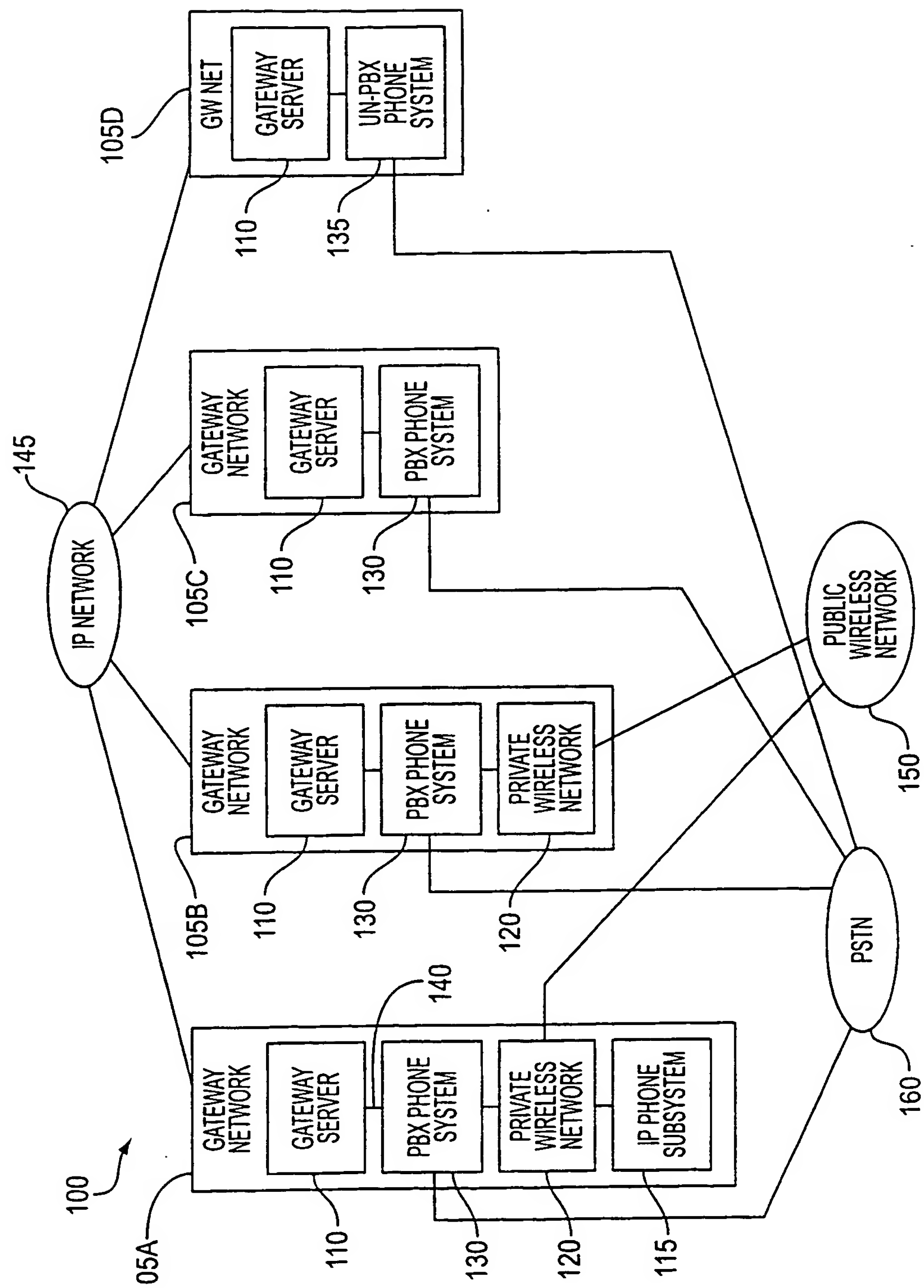


FIG. 1

BNSDOCID: <WO 0069156A1 I >

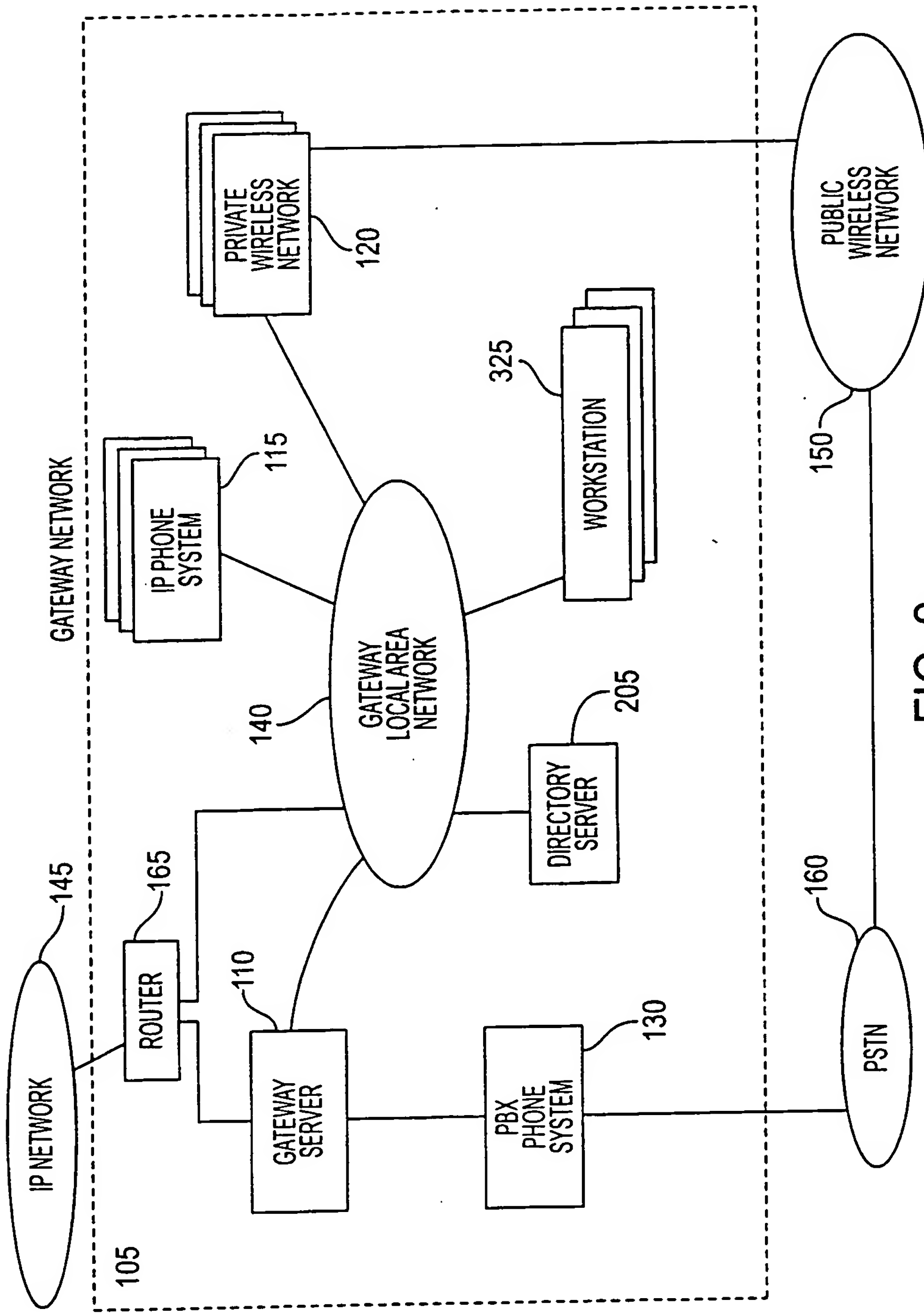

GATEWAY NETWORK
105
IP NETWORK
145
ROUTER
165
GATEWAY SERVER
110
PBX PHONE SYSTEM
130
PSTN
160
GATEWAY LOCAL AREA NETWORK
140
DIRECTORY SERVER
205
IP PHONE SYSTEM
115
PRIVATE WIRELESS NETWORK
120
WORKSTATION
325
PUBLIC WIRELESS NETWORK
150

FIG. 2

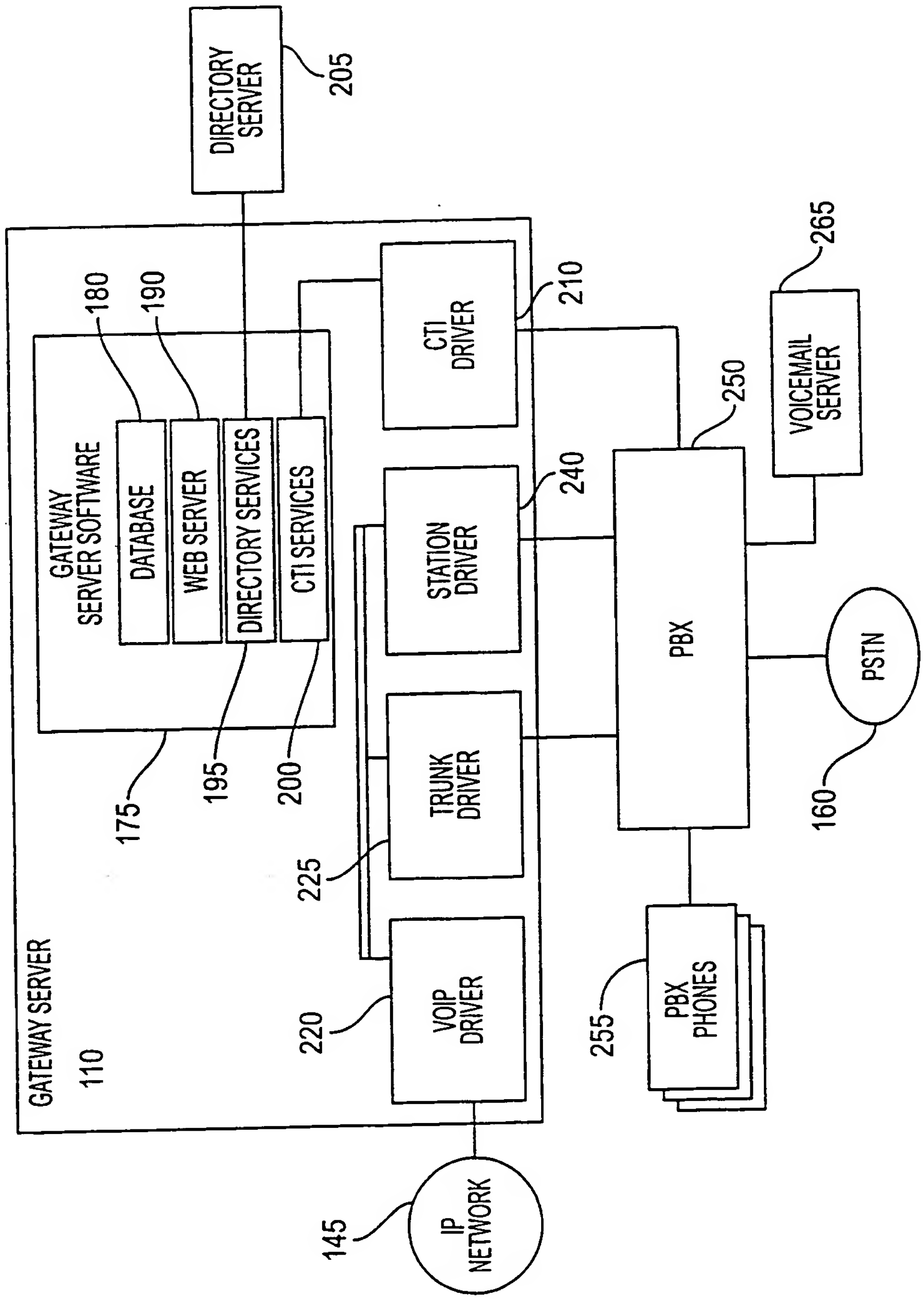


FIG. 3

BNSDOCID: <WO 0069156A1 I >

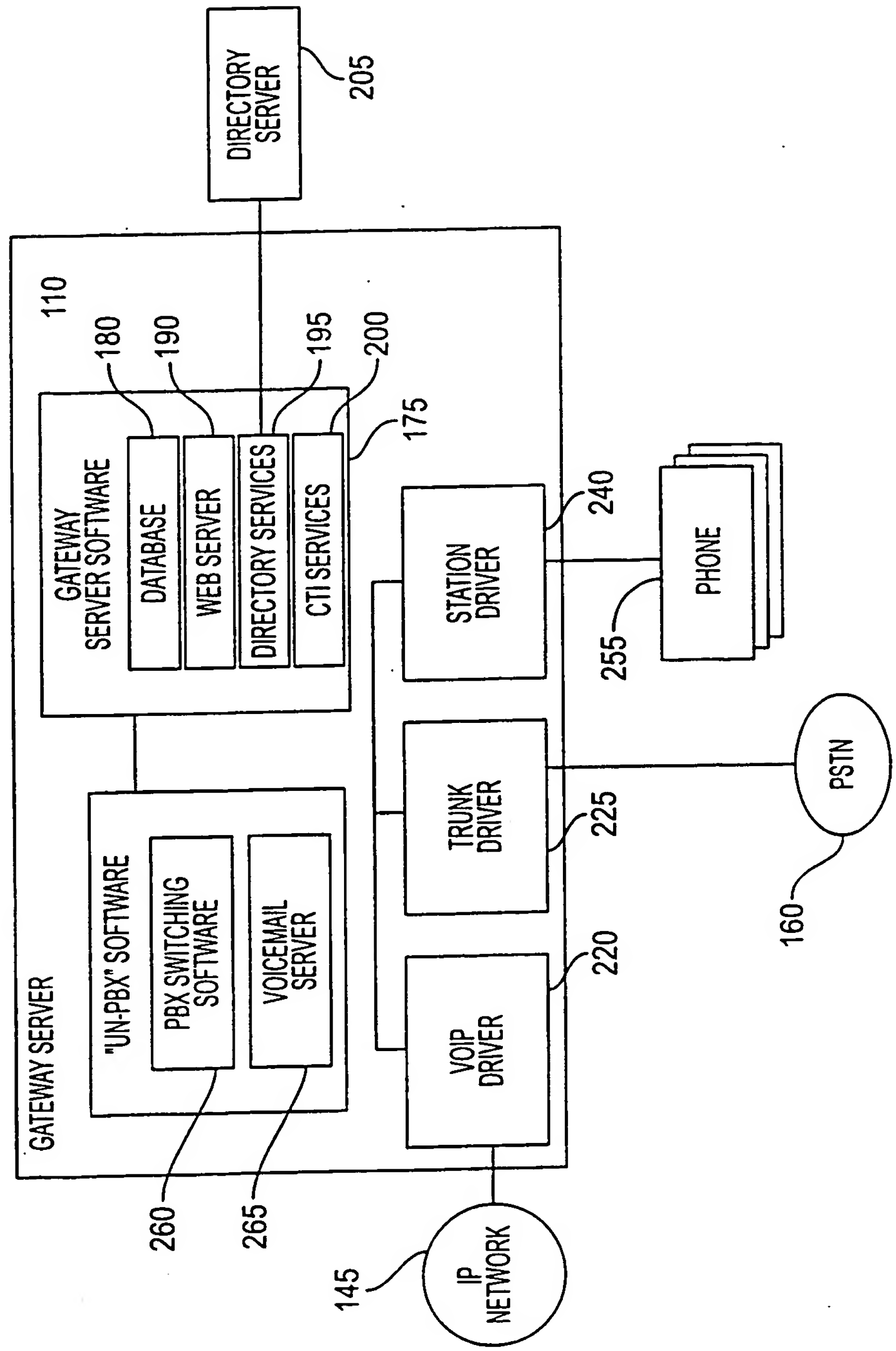

DIRECTORY SERVER
205
110
GATEWAY SERVER
GATEWAY SERVER SOFTWARE
175
DATABASE
180
WEB SERVER
190
DIRECTORY SERVICES
195
CTI SERVICES
200
"UN-PBX" SOFTWARE
PBX SWITCHING SOFTWARE
260
VOICEMAIL SERVER
265
STATION DRIVER
240
TRUNK DRIVER
225
VOIP DRIVER
220
PHONE
255
PSTN
160
IP NETWORK
145

FIG. 4

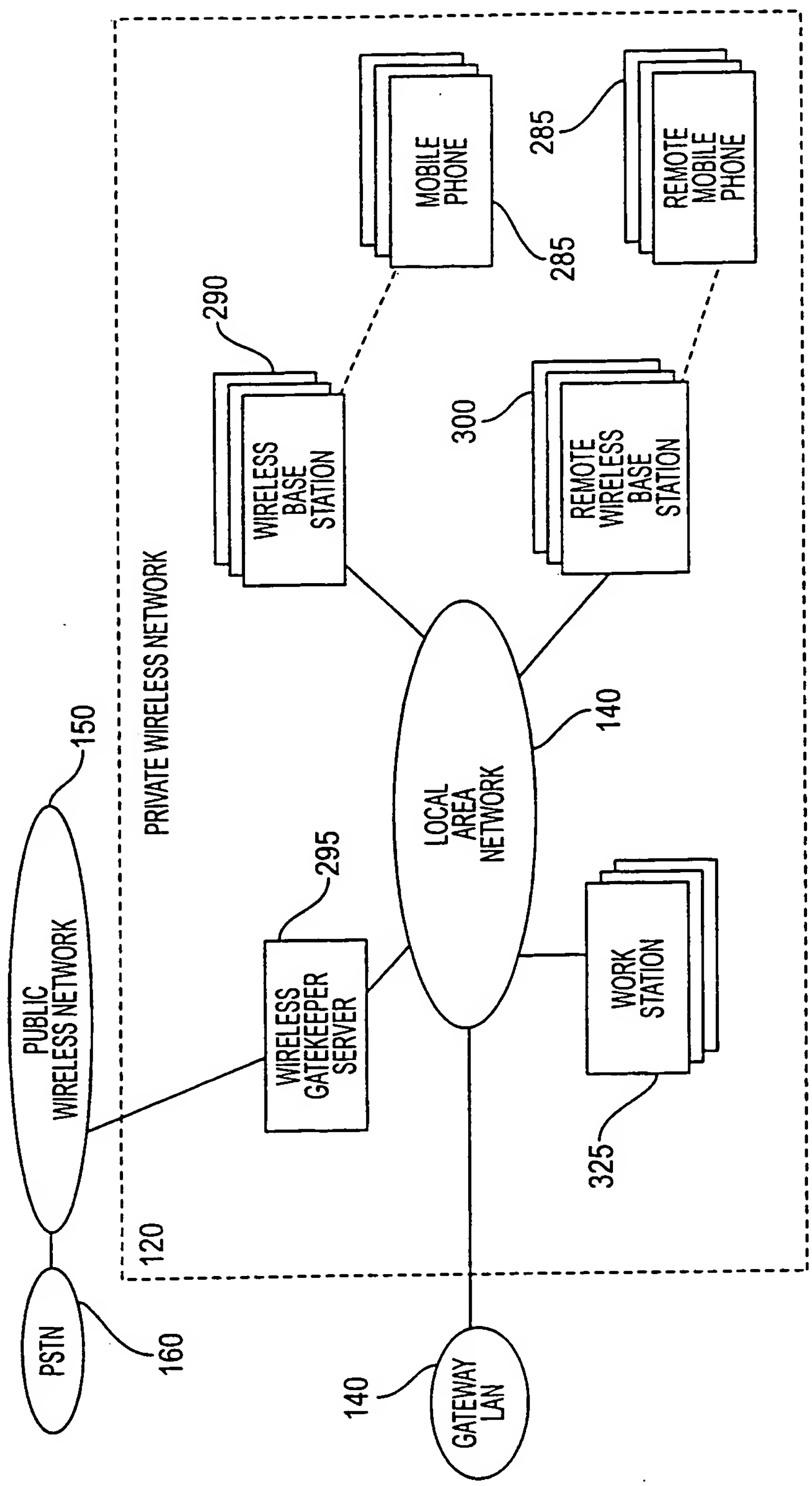

PRIVATE WIRELESS NETWORK
MOBILE PHONE
285
285
REMOTE MOBILE PHONE
290
WIRELESS BASE STATION
300
REMOTE WIRELESS BASE STATION
150
PUBLIC WIRELESS NETWORK
295
WIRELESS GATEKEEPER SERVER
LOCAL AREA NETWORK
140
WORK STATION
325
120
PSTN
160
140
GATEWAY LAN

FIG. 5

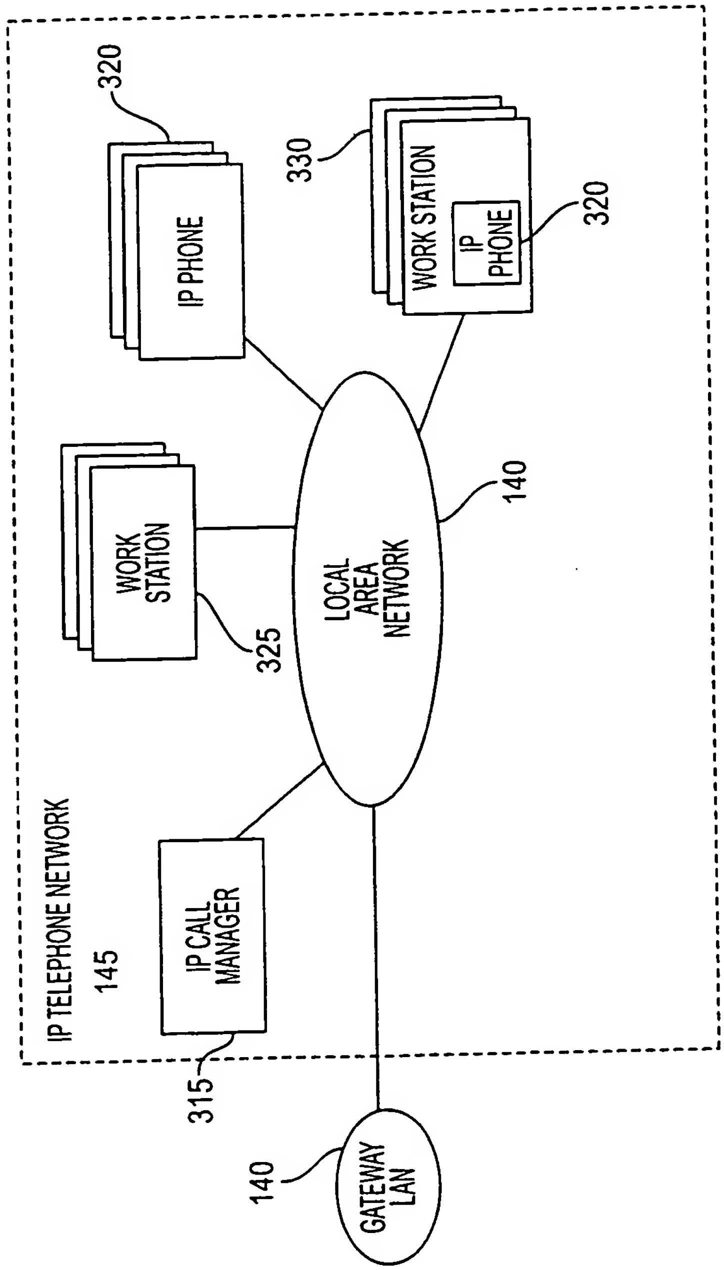

IP TELEPHONE NETWORK
145
IP CALL MANAGER
315
LOCAL AREA NETWORK
140
WORK STATION
325
IP PHONE
320
330
WORK STATION
IP PHONE
320
GATEWAY LAN
140

FIG. 6

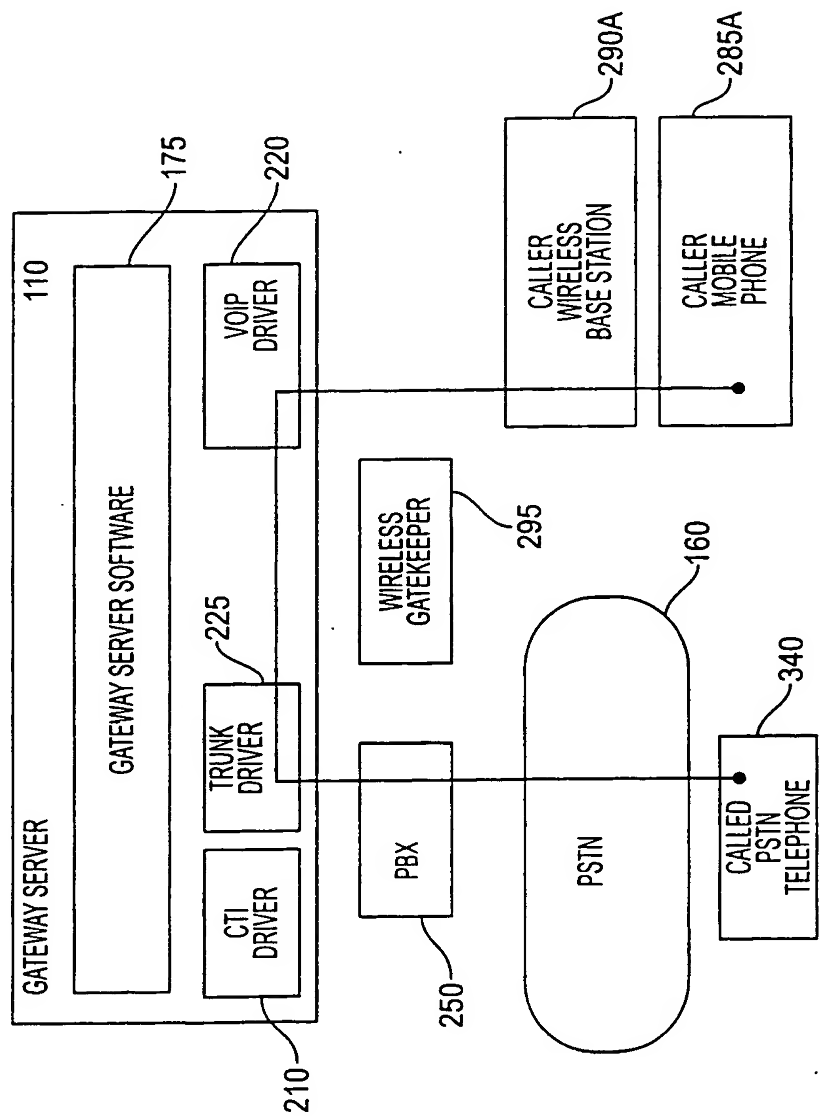


FIG. 7

BNSDOCID: <WO 0069156A1 I >

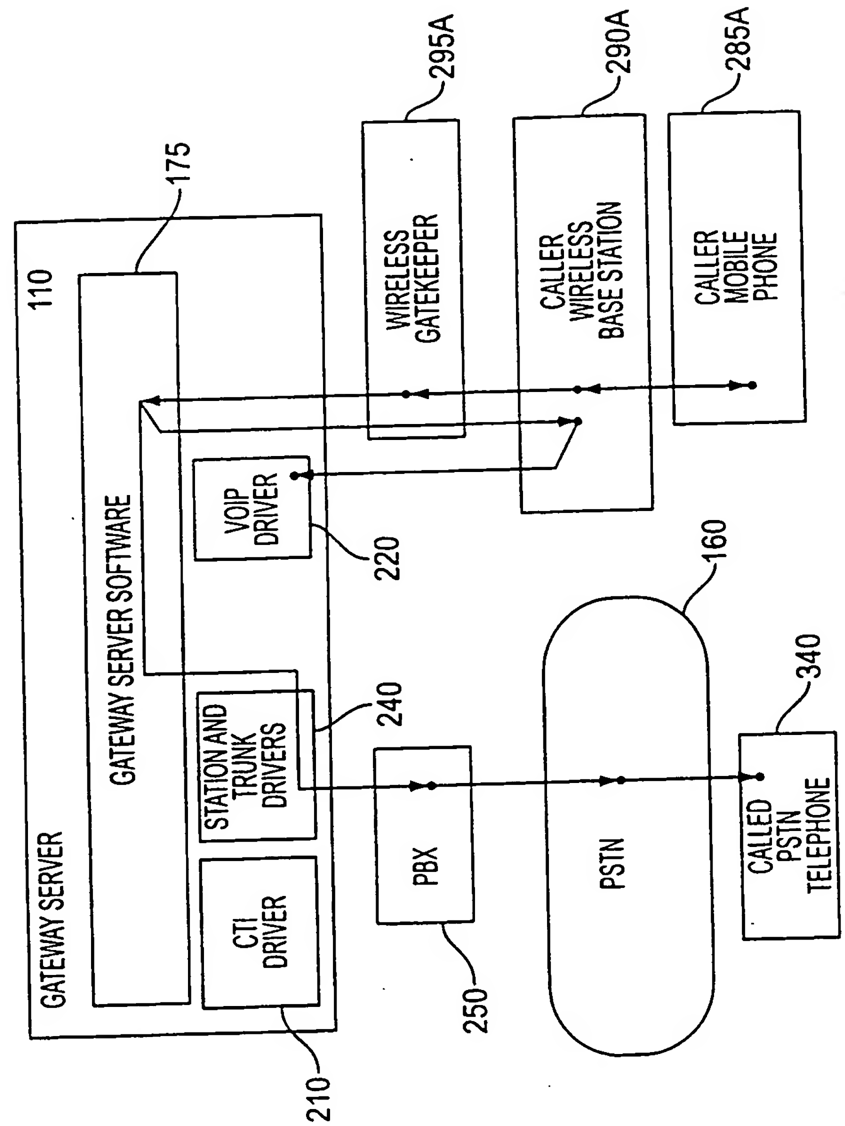


FIG. 8

BNSDOCID: <WO 0069156A1 I >

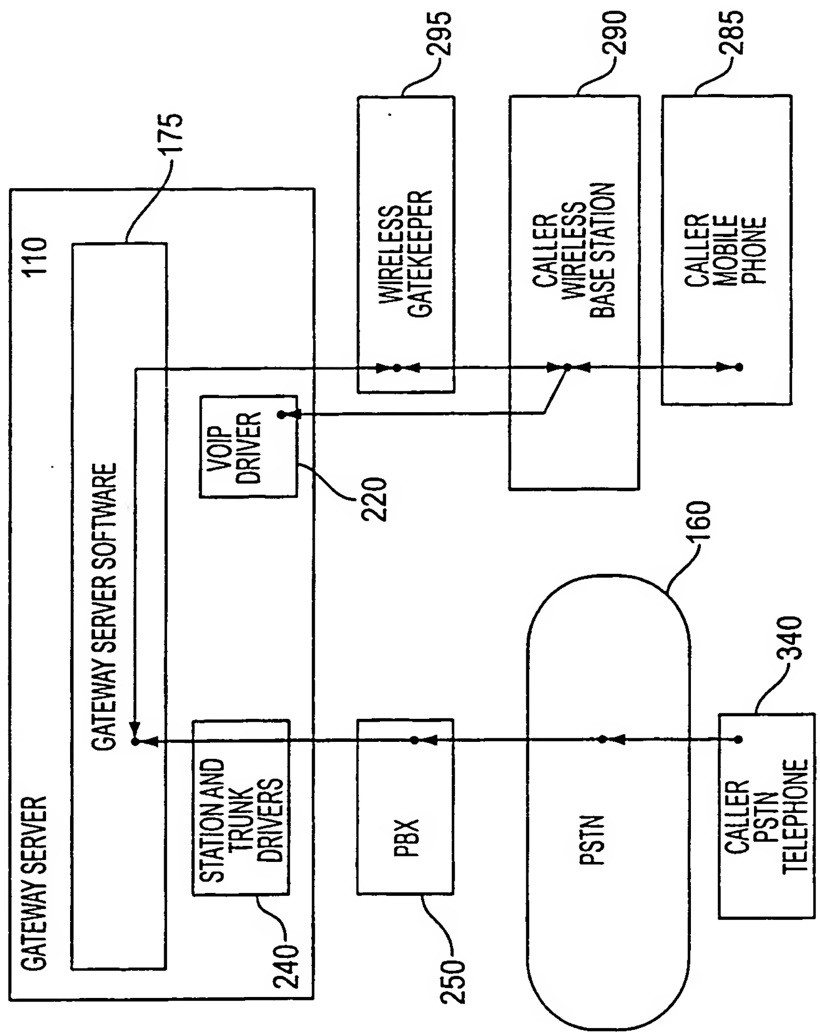

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
175
VOIP DRIVER
220
STATION AND TRUNK DRIVERS
240
PBX
250
PSTN
160
WIRELESS GATEKEEPER
295
CALLER WIRELESS BASE STATION
290
CALLER MOBILE PHONE
285
CALLER PSTN TELEPHONE
340

FIG. 9

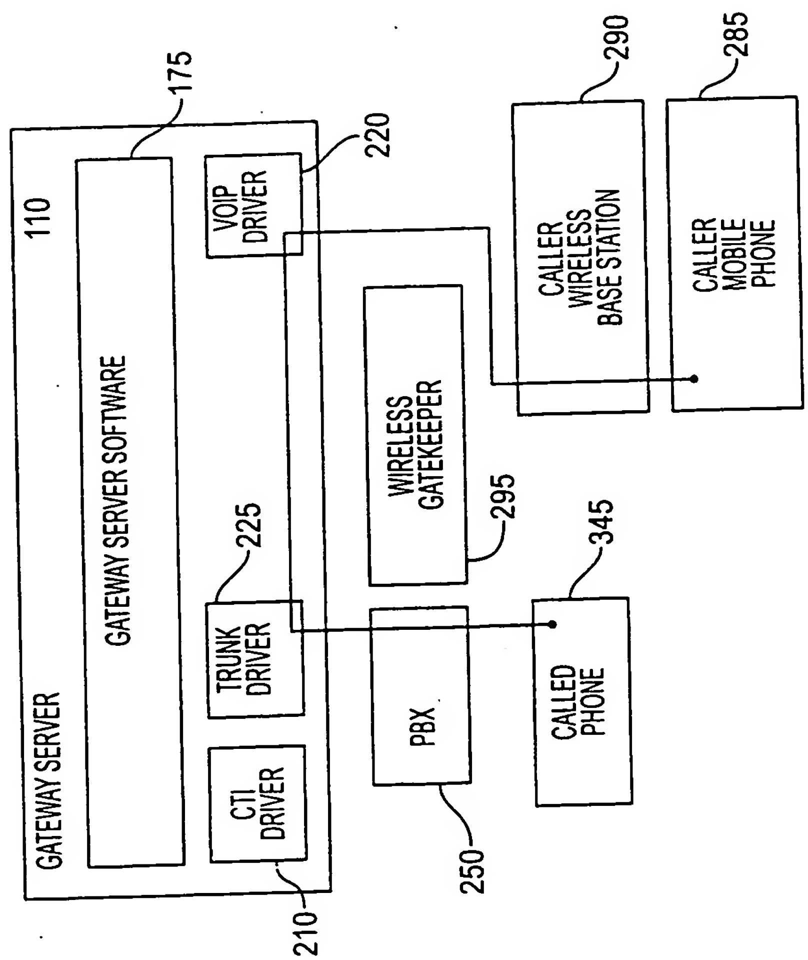

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
175
VOIP DRIVER
220
TRUNK DRIVER
225
CTI DRIVER
210
WIRELESS GATEKEEPER
295
PBX
250
CALLED PHONE
345
CALLER WIRELESS BASE STATION
290
CALLER MOBILE PHONE
285

FIG. 10

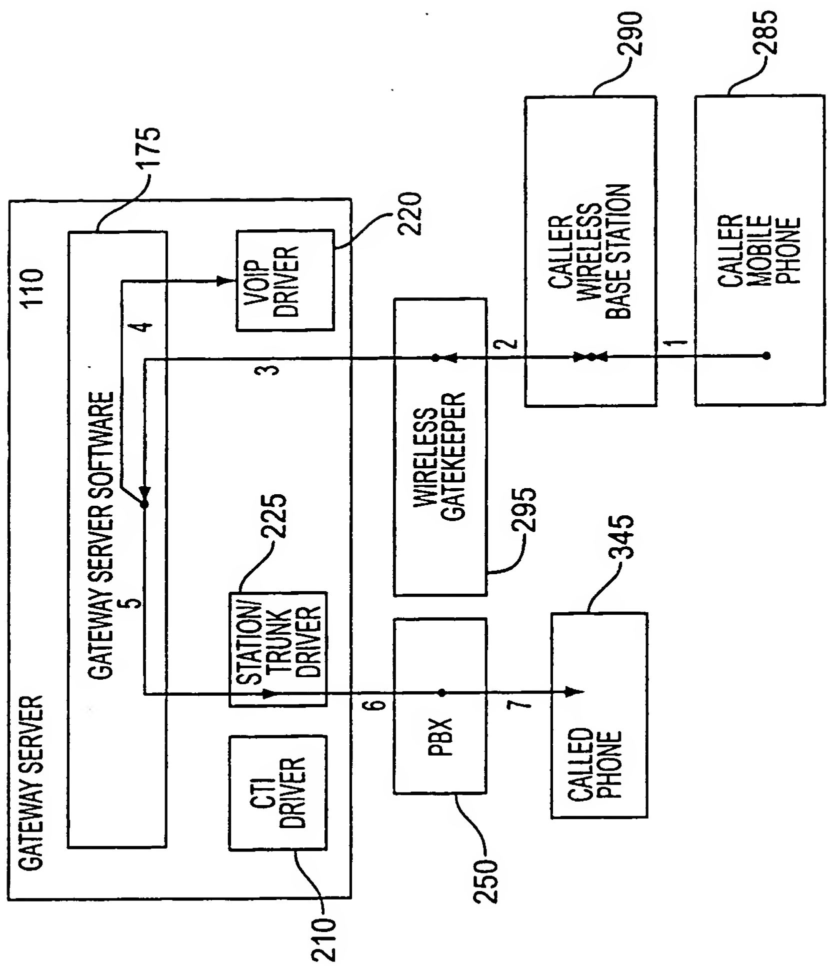

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
175
VOIP DRIVER
220
STATION/ TRUNK DRIVER
225
CTI DRIVER
210
WIRELESS GATEKEEPER
295
PBX
250
CALLER WIRELESS BASE STATION
290
CALLER MOBILE PHONE
285
CALLED PHONE
345
1
2
3
4
5
6
7

FIG. 11

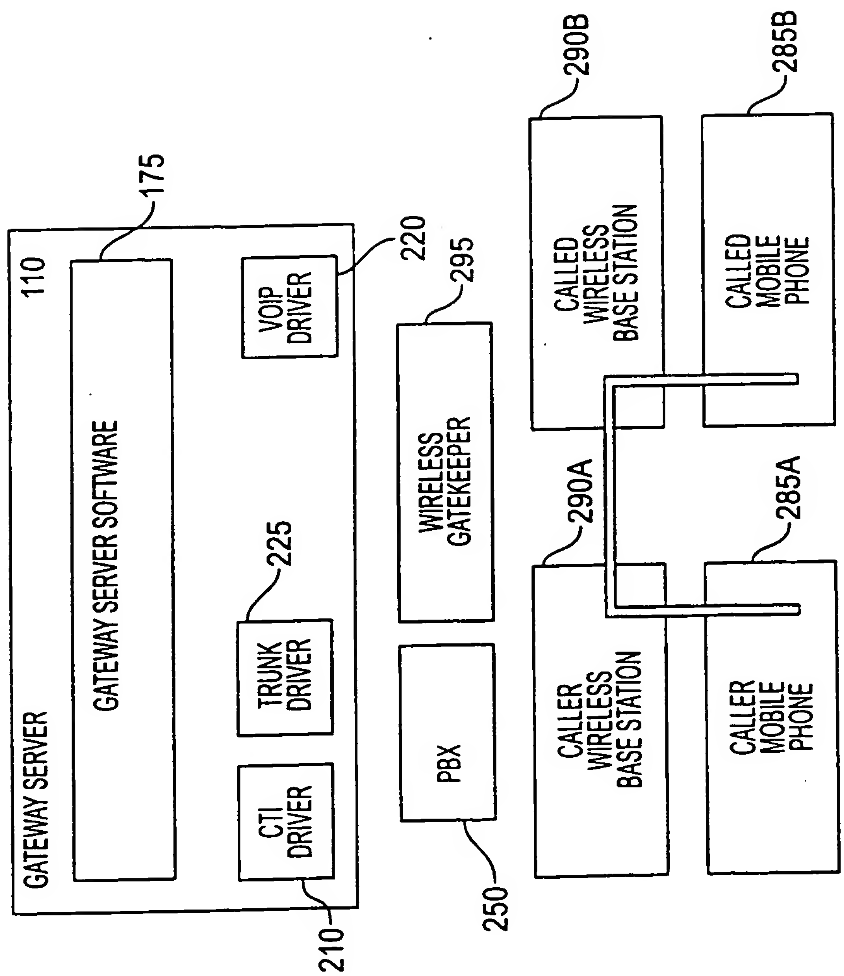

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
175
CTI DRIVER
210
TRUNK DRIVER
225
VOIP DRIVER
220
PBX
250
WIRELESS GATEKEEPER
295
CALLER WIRELESS BASE STATION
290A
CALLED WIRELESS BASE STATION
290B
CALLER MOBILE PHONE
285A
CALLED MOBILE PHONE
285B

FIG. 12

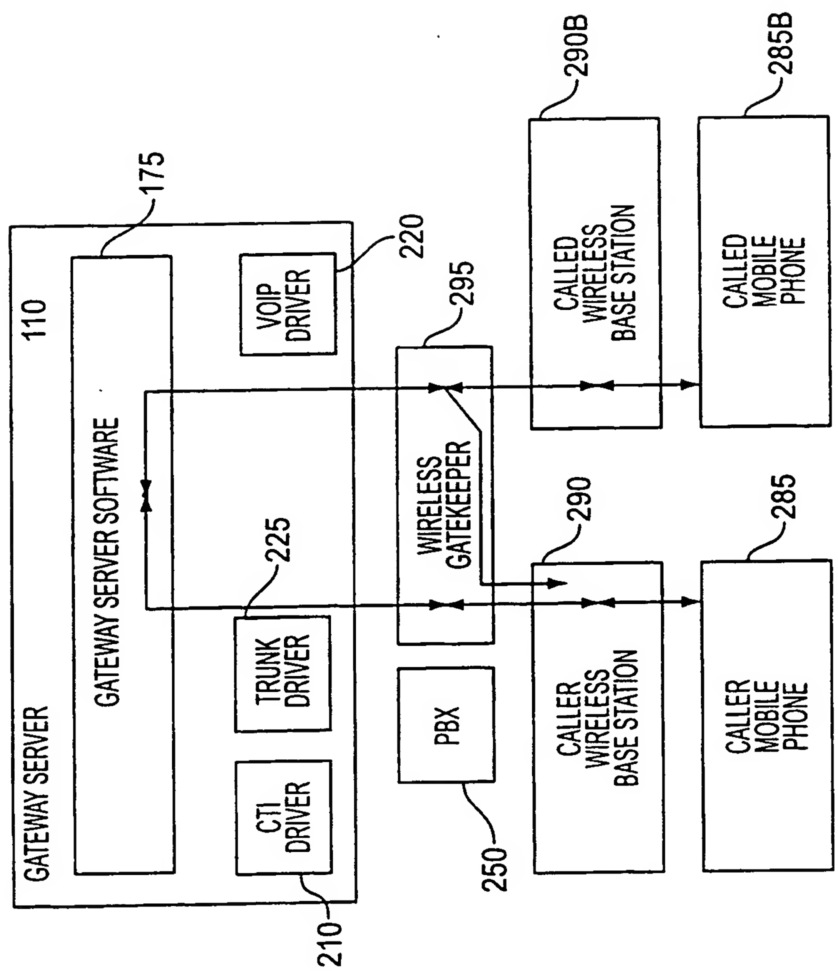

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
175
VOIP DRIVER
220
TRUNK DRIVER
225
CTI DRIVER
210
WIRELESS GATEKEEPER
295
PBX
250
CALLED WIRELESS BASE STATION
290B
CALLER WIRELESS BASE STATION
290
CALLED MOBILE PHONE
285B
CALLER MOBILE PHONE
285

FIG. 13

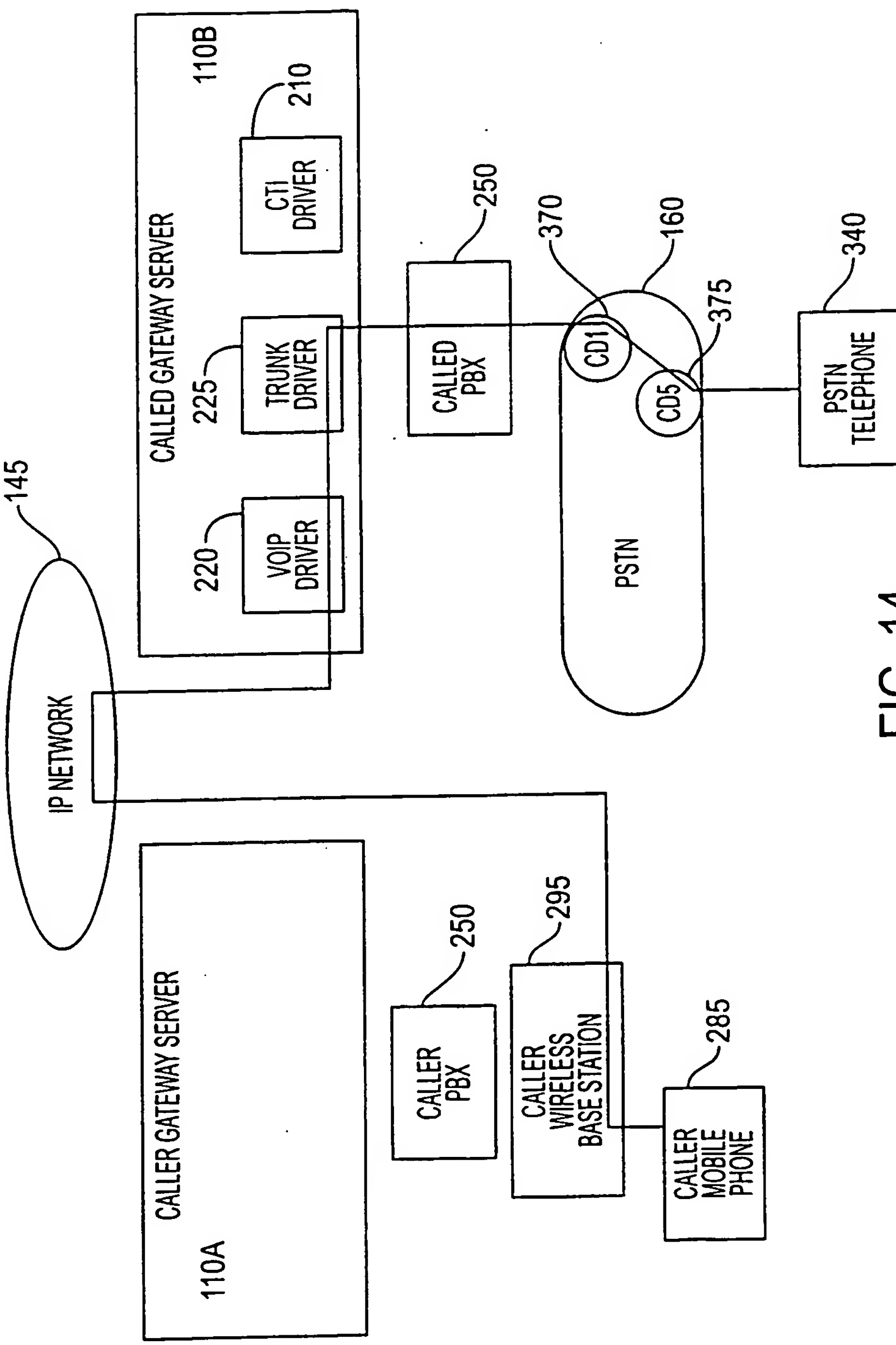

CALLED GATEWAY SERVER
110B
CTI DRIVER
210
TRUNK DRIVER
225
VOIP DRIVER
220
145
IP NETWORK
CALLED PBX
250
370
160
375
CD1
CD5
PSTN
PSTN TELEPHONE
340
CALLER GATEWAY SERVER
110A
CALLER PBX
250
CALLER WIRELESS BASE STATION
295
CALLER MOBILE PHONE
285

FIG. 14

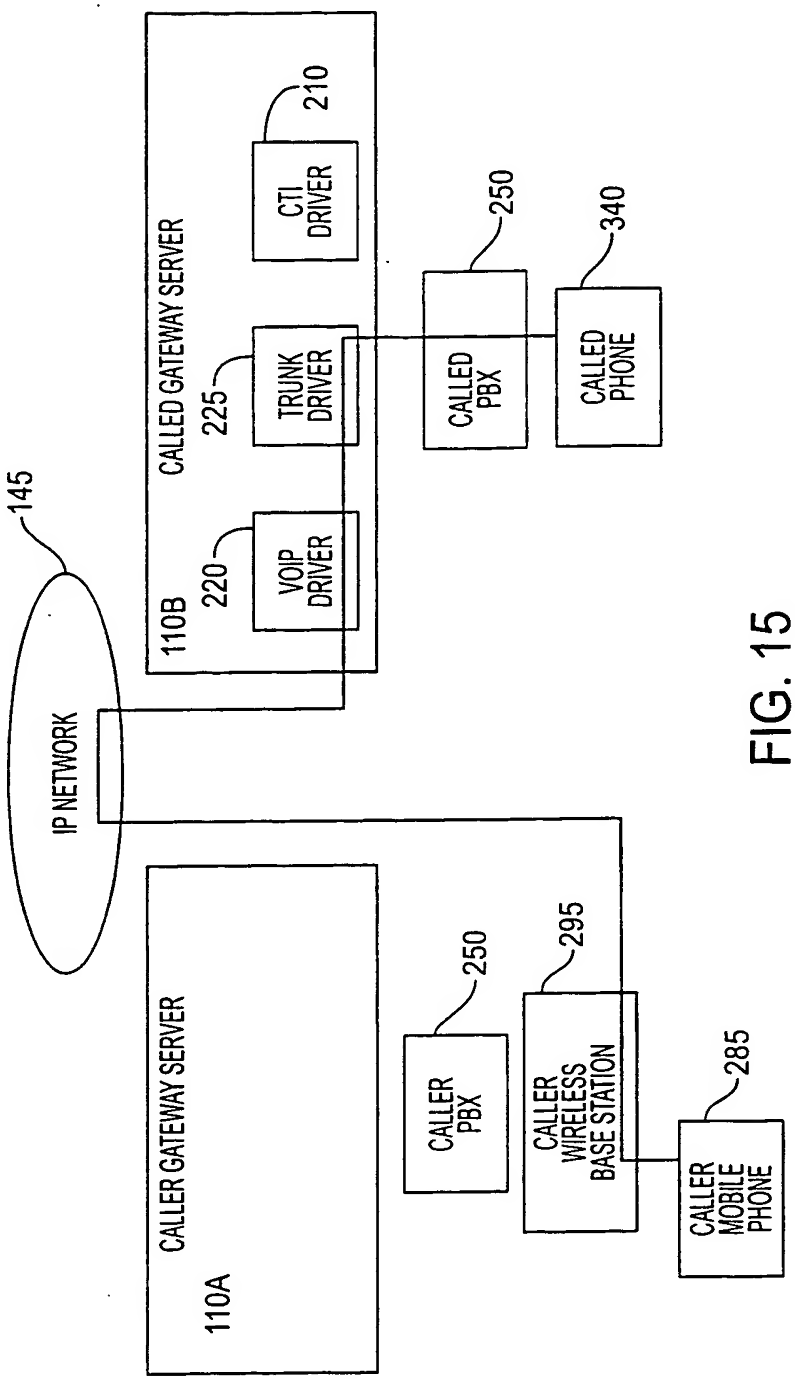

CALLED GATEWAY SERVER
110B
CTI DRIVER
210
TRUNK DRIVER
225
VOIP DRIVER
220
CALLED PBX
250
CALLED PHONE
340
IP NETWORK
145
CALLER GATEWAY SERVER
110A
CALLER PBX
250
CALLER WIRELESS BASE STATION
295
CALLER MOBILE PHONE
285

FIG. 15

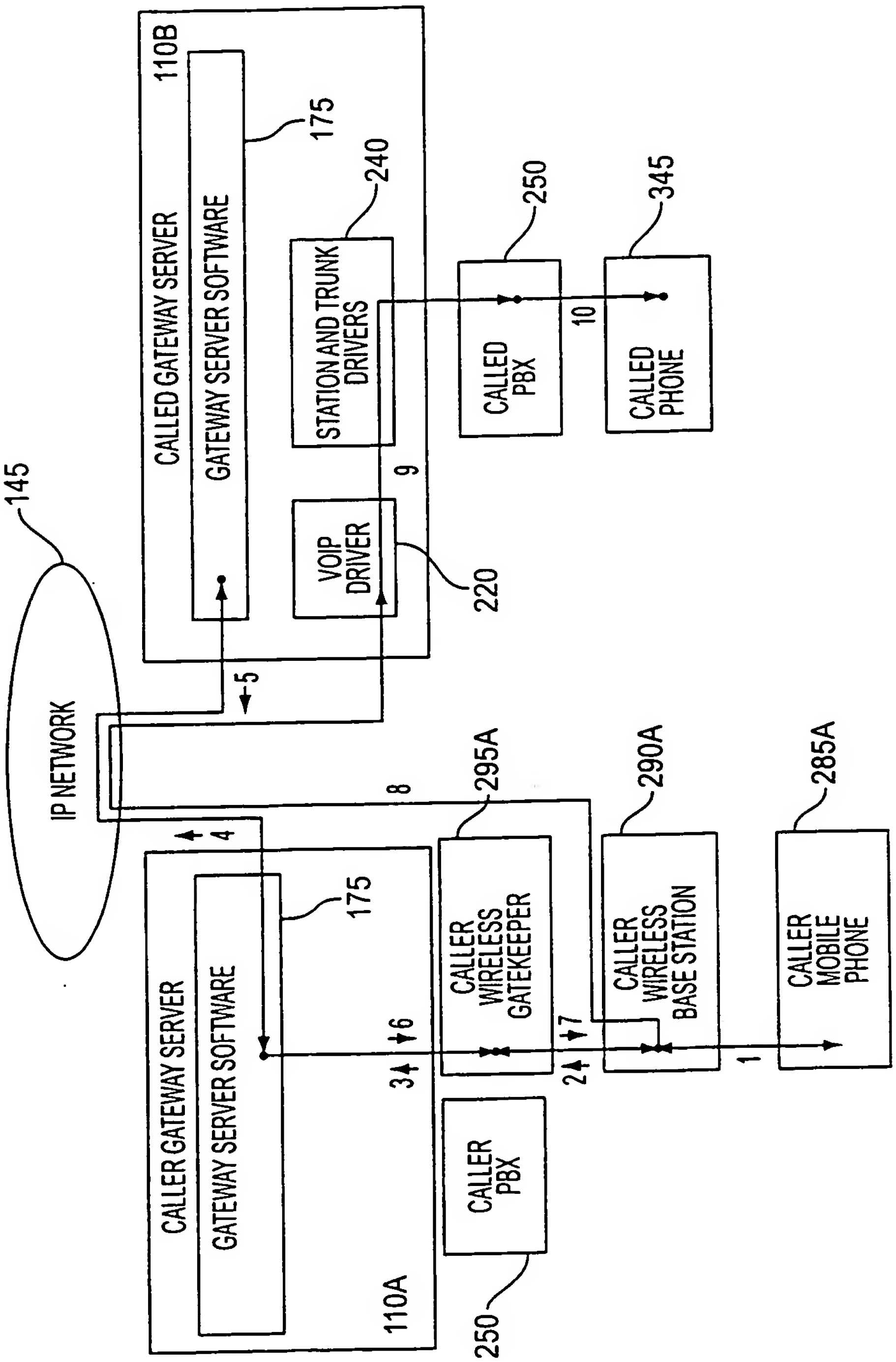

CALLED GATEWAY SERVER
110B
GATEWAY SERVER SOFTWARE
175
STATION AND TRUNK DRIVERS
240
VOIP DRIVER
220
CALLED PBX
250
CALLED PHONE
345
IP NETWORK
145
CALLER GATEWAY SERVER
GATEWAY SERVER SOFTWARE
175
110A
CALLER WIRELESS GATEKEEPER
295A
CALLER WIRELESS BASE STATION
290A
CALLER MOBILE PHONE
285A
CALLER PBX
250
1
2
3
4
5
6
7
8
9
10

FIG. 16

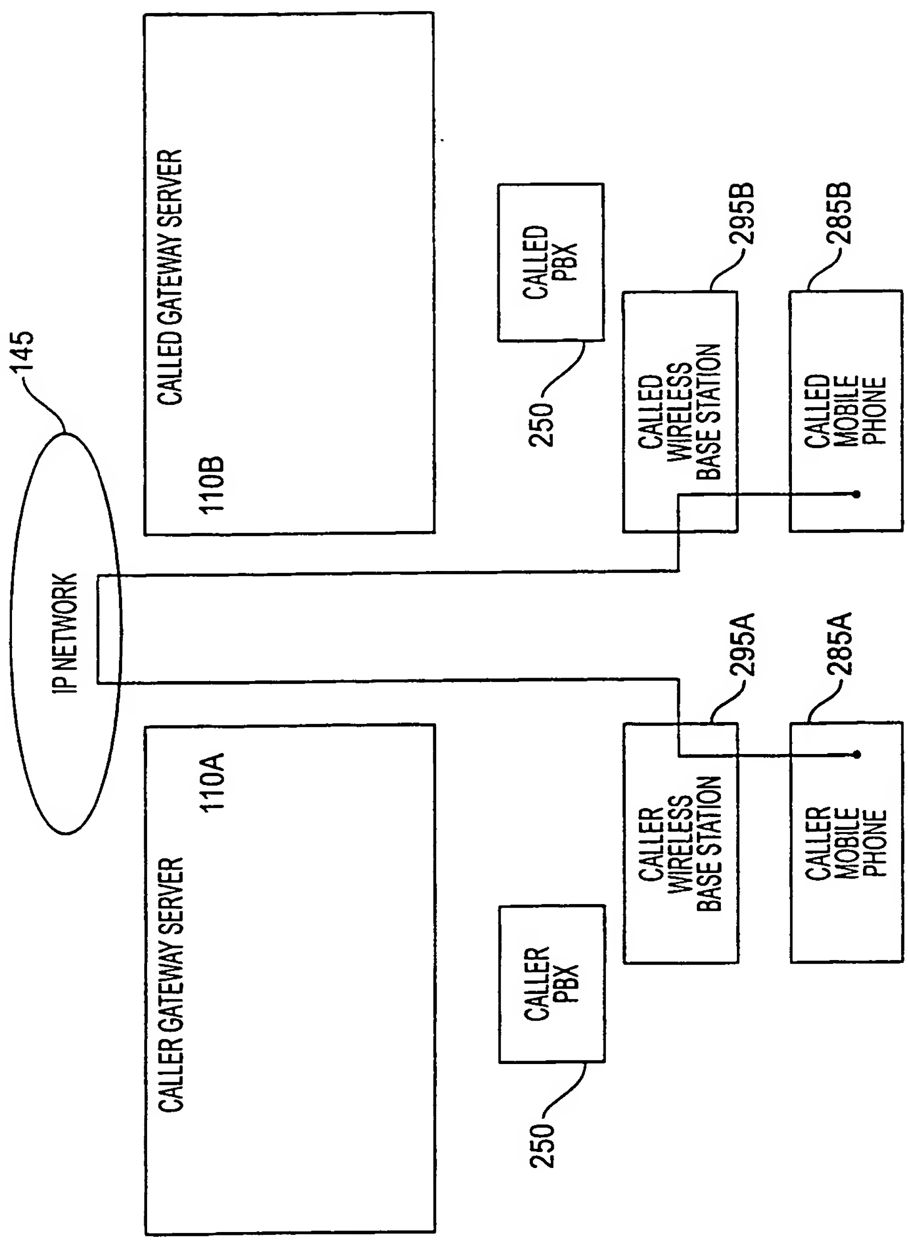

145
IP NETWORK
CALLED GATEWAY SERVER
110B
CALLER GATEWAY SERVER
110A
CALLED PBX
250
CALLER PBX
250
CALLED WIRELESS BASE STATION
295B
CALLER WIRELESS BASE STATION
295A
CALLED MOBILE PHONE
285B
CALLER MOBILE PHONE
285A

FIG. 17

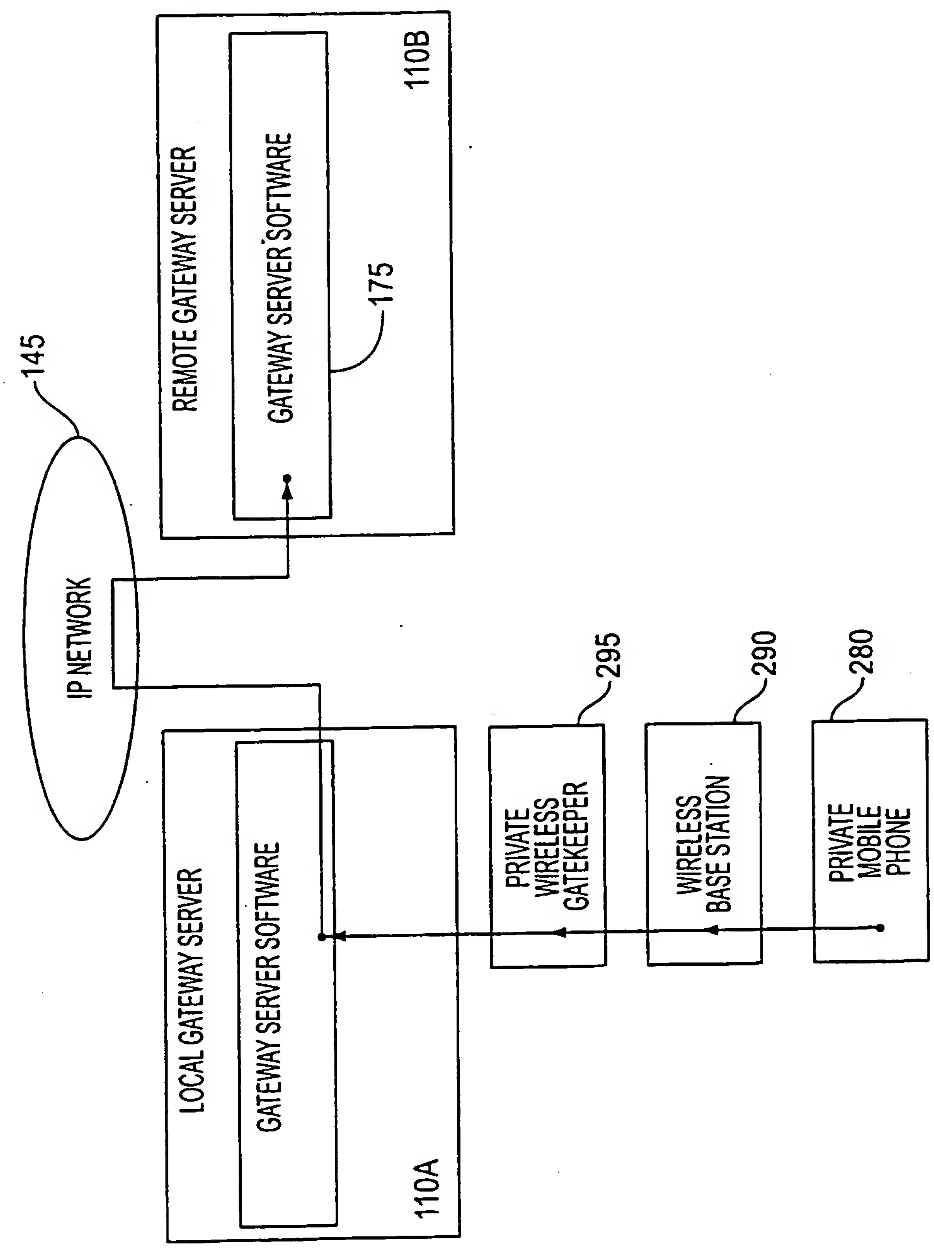

REMOTE GATEWAY SERVER
GATEWAY SERVER SOFTWARE
175
110B
145
IP NETWORK
LOCAL GATEWAY SERVER
GATEWAY SERVER SOFTWARE
110A
PRIVATE WIRELESS GATEKEEPER
295
WIRELESS BASE STATION
290
PRIVATE MOBILE PHONE
280

FIG. 18

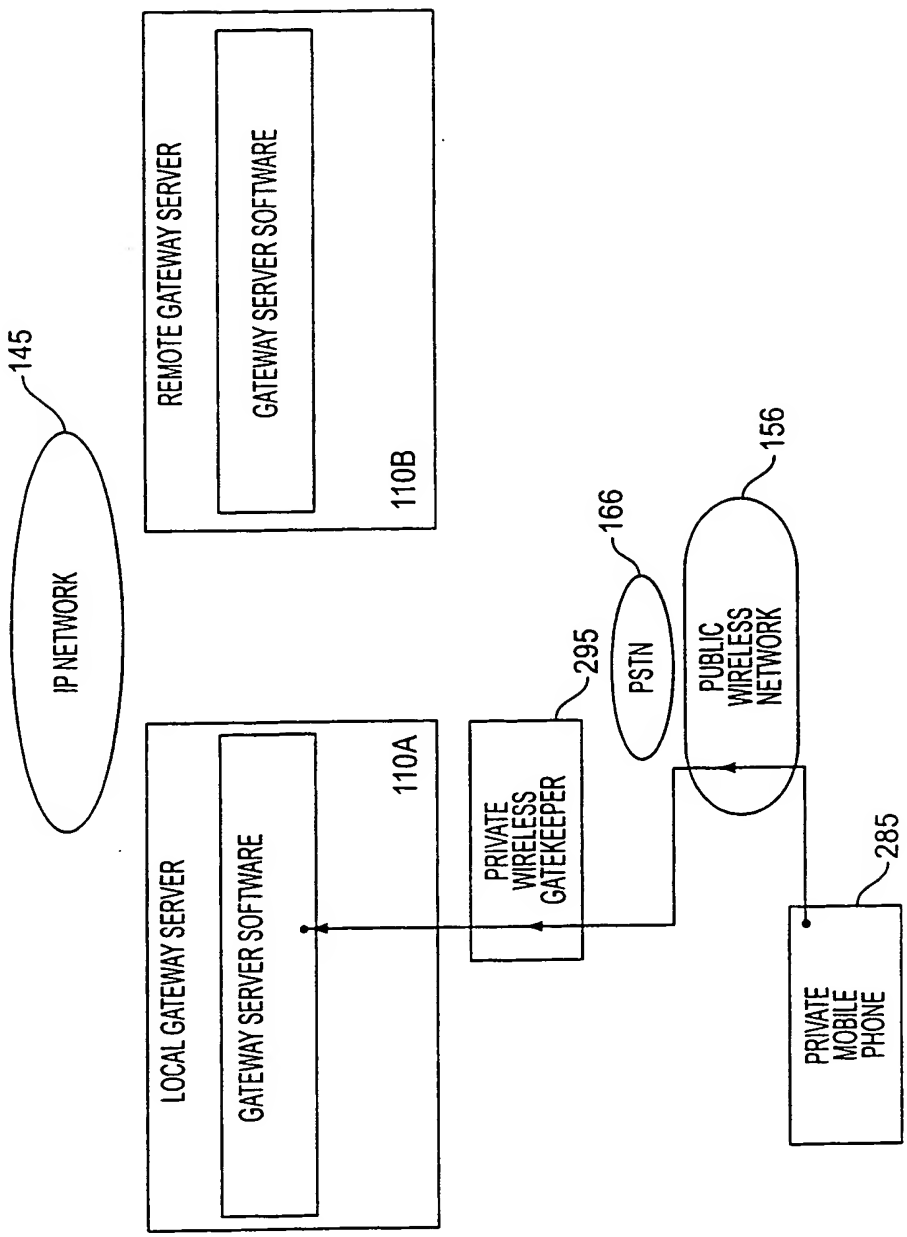

IP NETWORK
145
REMOTE GATEWAY SERVER
GATEWAY SERVER SOFTWARE
110B
LOCAL GATEWAY SERVER
GATEWAY SERVER SOFTWARE
110A
PRIVATE WIRELESS GATEKEEPER
295
PSTN
166
PUBLIC WIRELESS NETWORK
156
PRIVATE MOBILE PHONE
285

FIG. 19

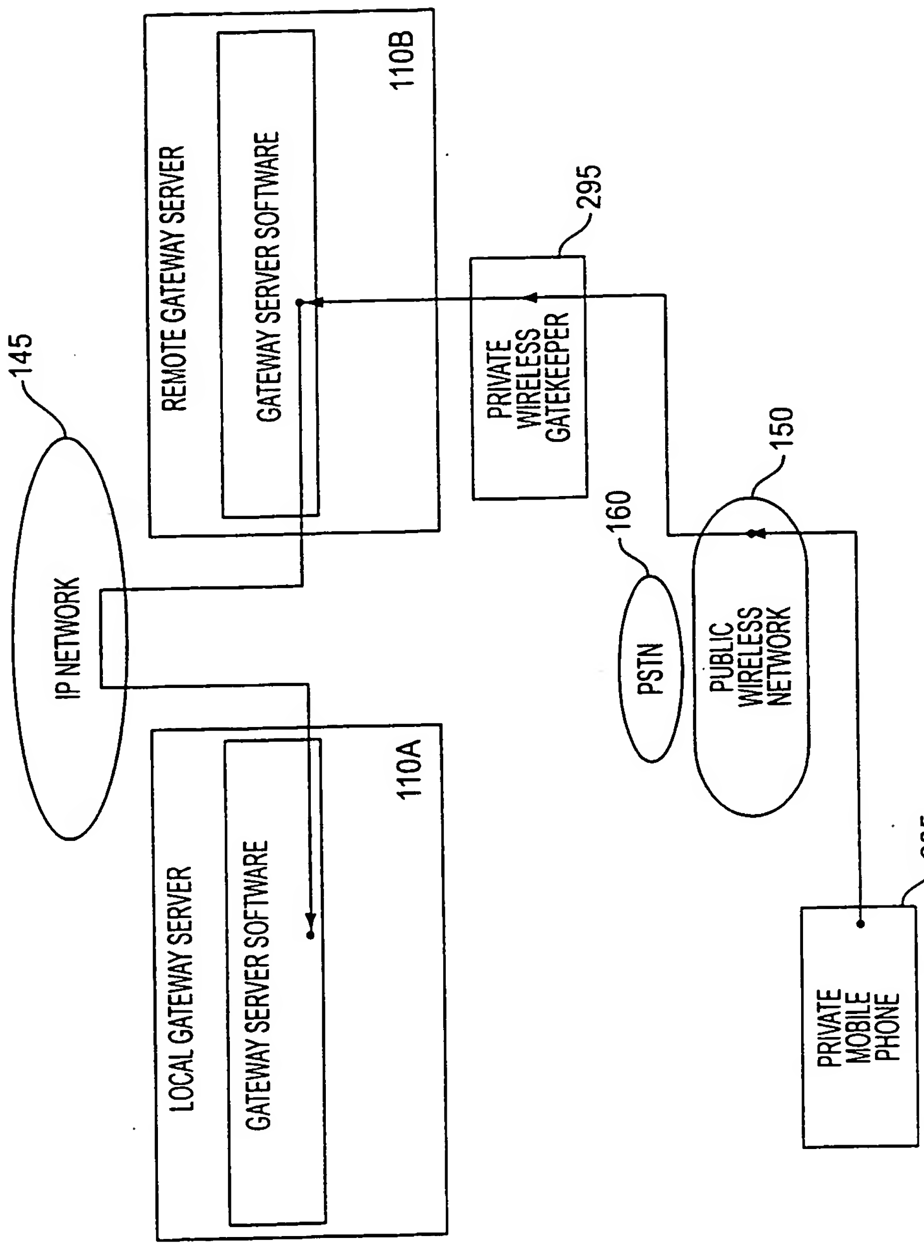


FIG. 20

BNSDOCID: <WO 0069156A1 I >

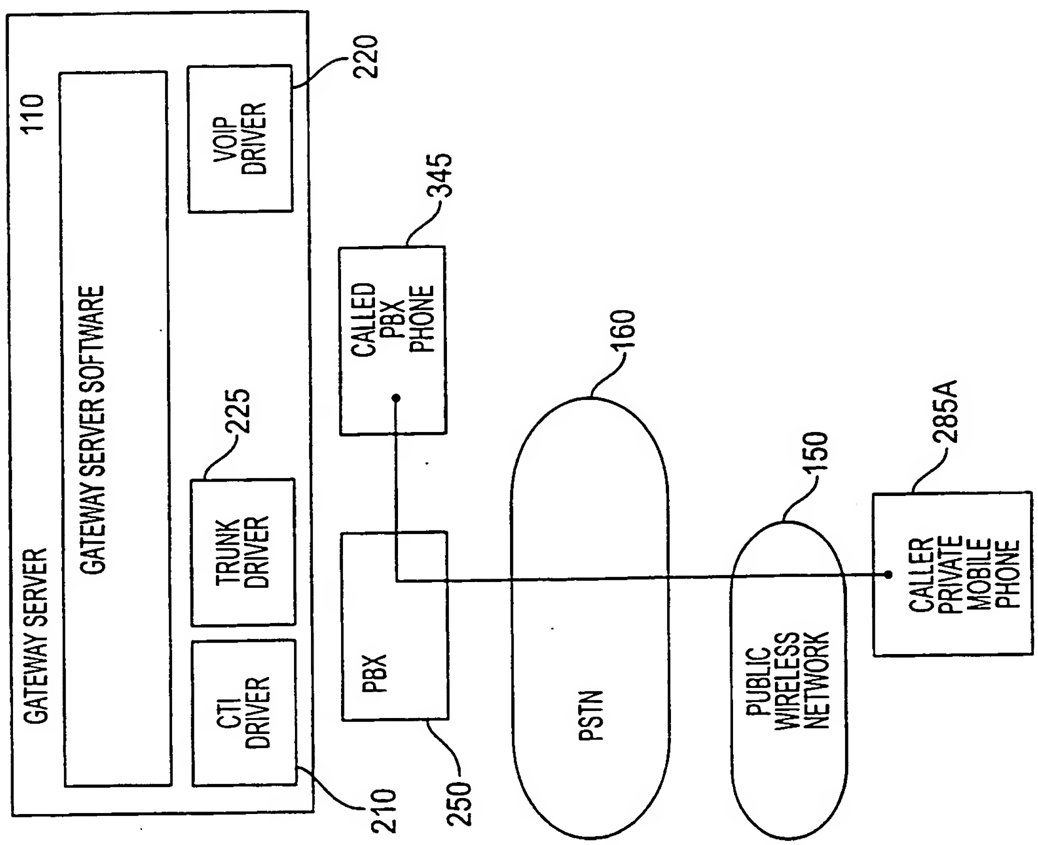

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
VOIP DRIVER
220
TRUNK DRIVER
225
CTI DRIVER
210
CALLED PBX PHONE
345
PBX
250
PSTN
160
PUBLIC WIRELESS NETWORK
150
CALLER PRIVATE MOBILE PHONE
285A

FIG. 21

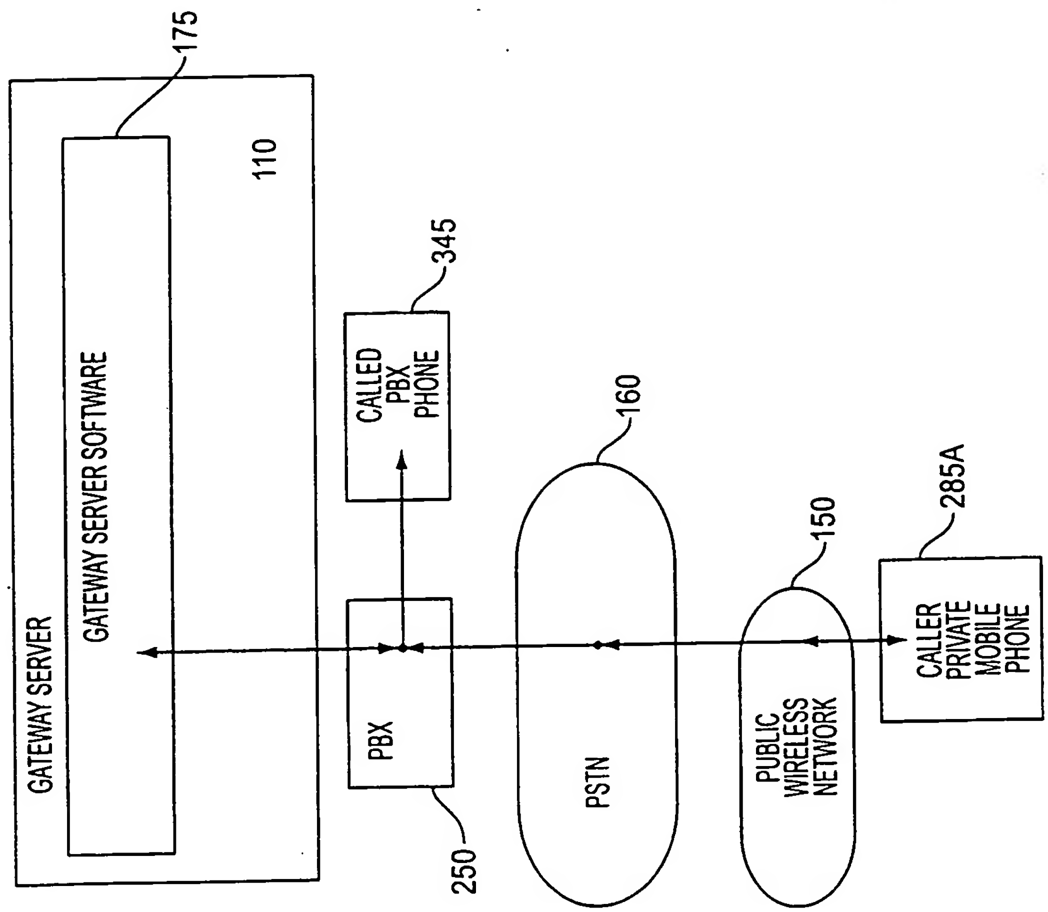


FIG. 22

BNSDOCID: <WO 0069156A1 I >

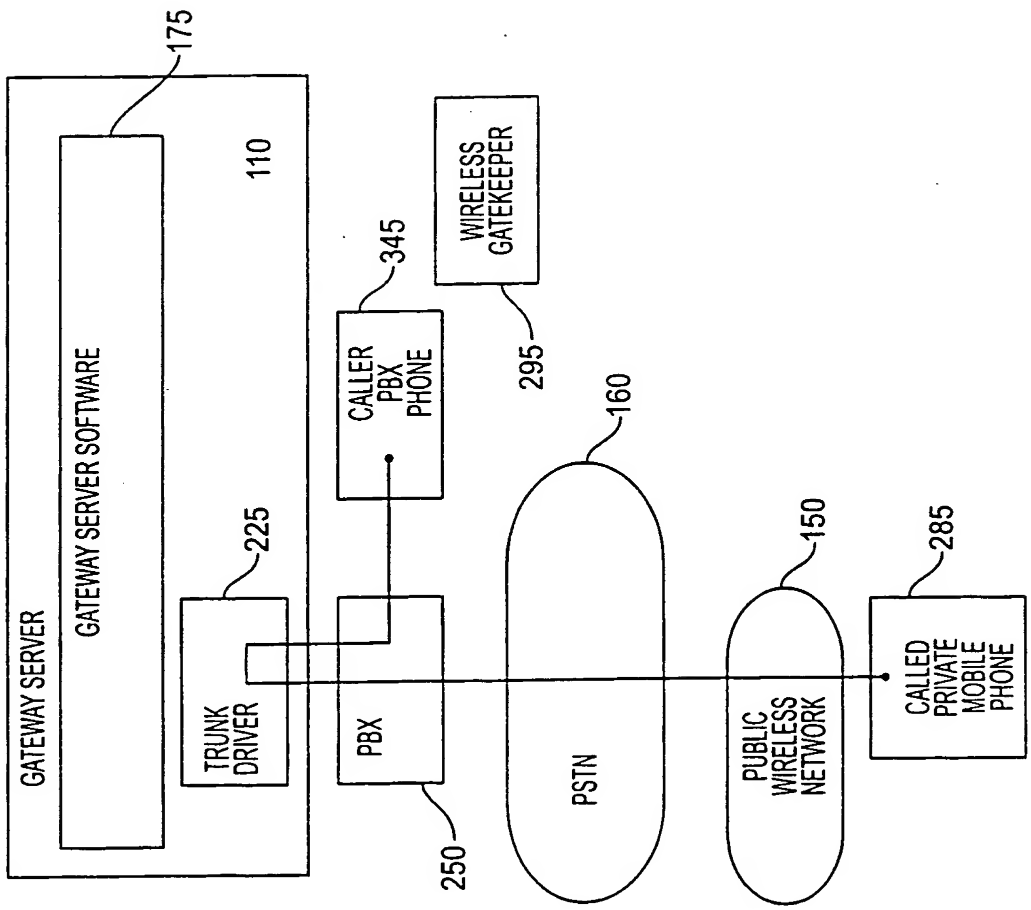

GATEWAY SERVER
GATEWAY SERVER SOFTWARE
175
110
TRUNK DRIVER
225
CALLER PBX PHONE
345
WIRELESS GATEKEEPER
295
PBX
250
PSTN
160
PUBLIC WIRELESS NETWORK
150
CALLED PRIVATE MOBILE PHONE
285

FIG. 23

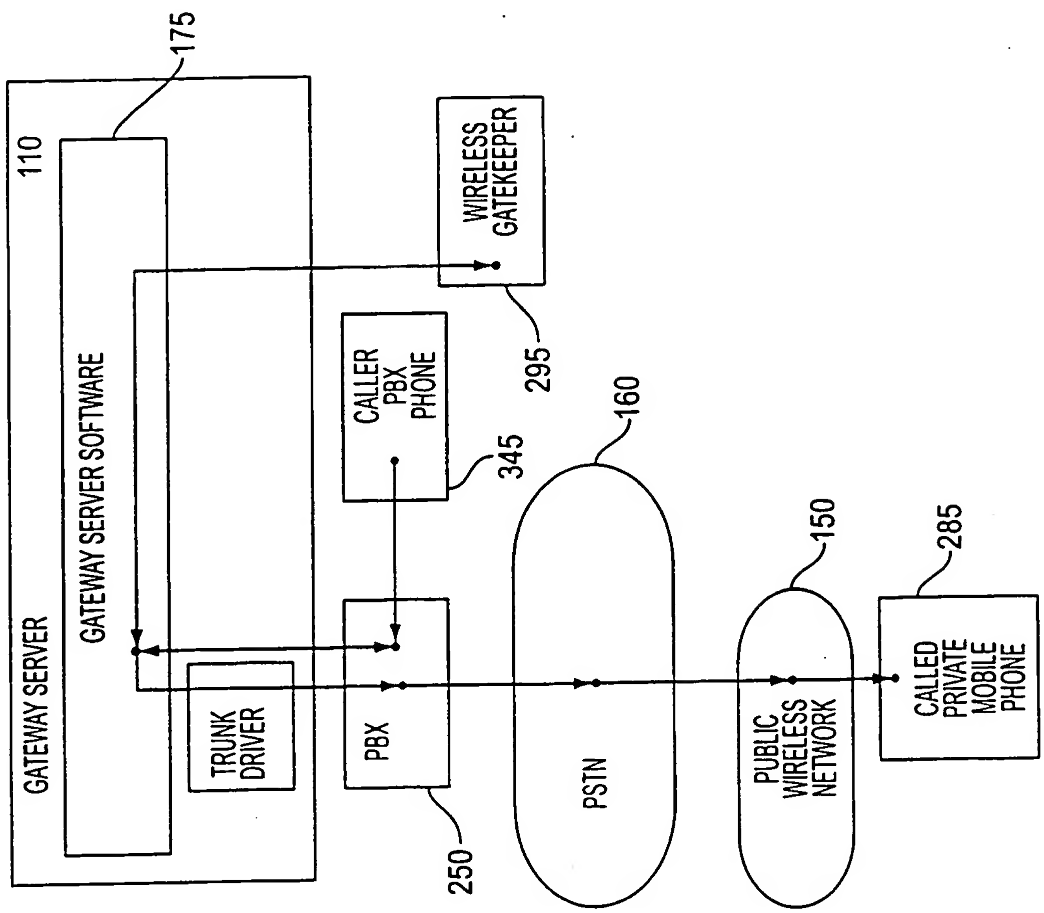

GATEWAY SERVER
110
GATEWAY SERVER SOFTWARE
175
TRUNK DRIVER
WIRELESS GATEKEEPER
295
CALLER PBX PHONE
345
PBX
250
PSTN
160
PUBLIC WIRELESS NETWORK
150
CALLED PRIVATE MOBILE PHONE
285

FIG. 24

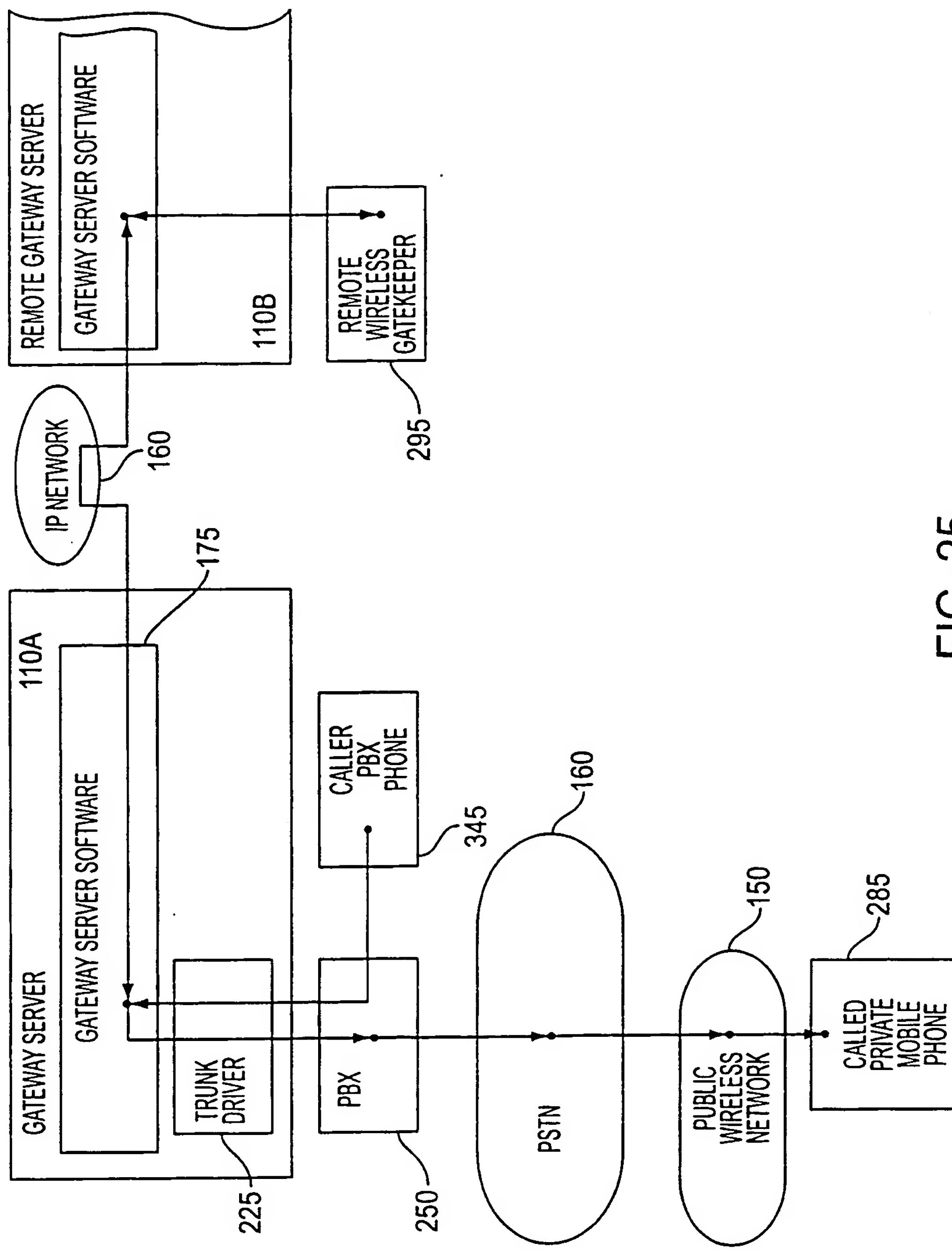

REMOTE GATEWAY SERVER
GATEWAY SERVER SOFTWARE
110B
REMOTE WIRELESS GATEKEEPER
295
IP NETWORK
160
175
110A
GATEWAY SERVER
GATEWAY SERVER SOFTWARE
TRUNK DRIVER
225
CALLER PBX PHONE
345
PBX
250
160
PSTN
150
PUBLIC WIRELESS NETWORK
285
CALLED PRIVATE MOBILE PHONE

FIG. 25

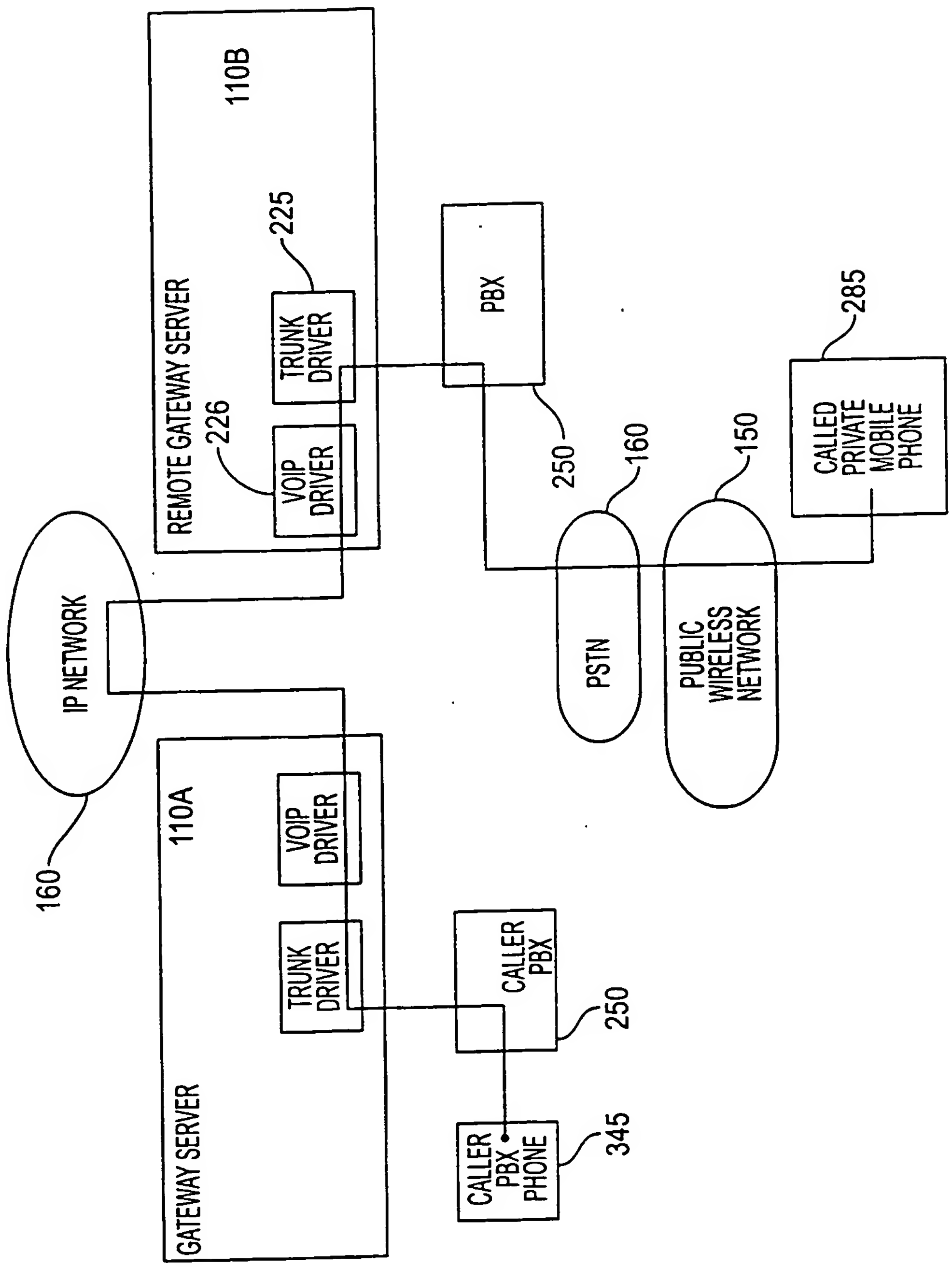


FIG. 26

BNSDOCID: <WO 0069156A1 I >

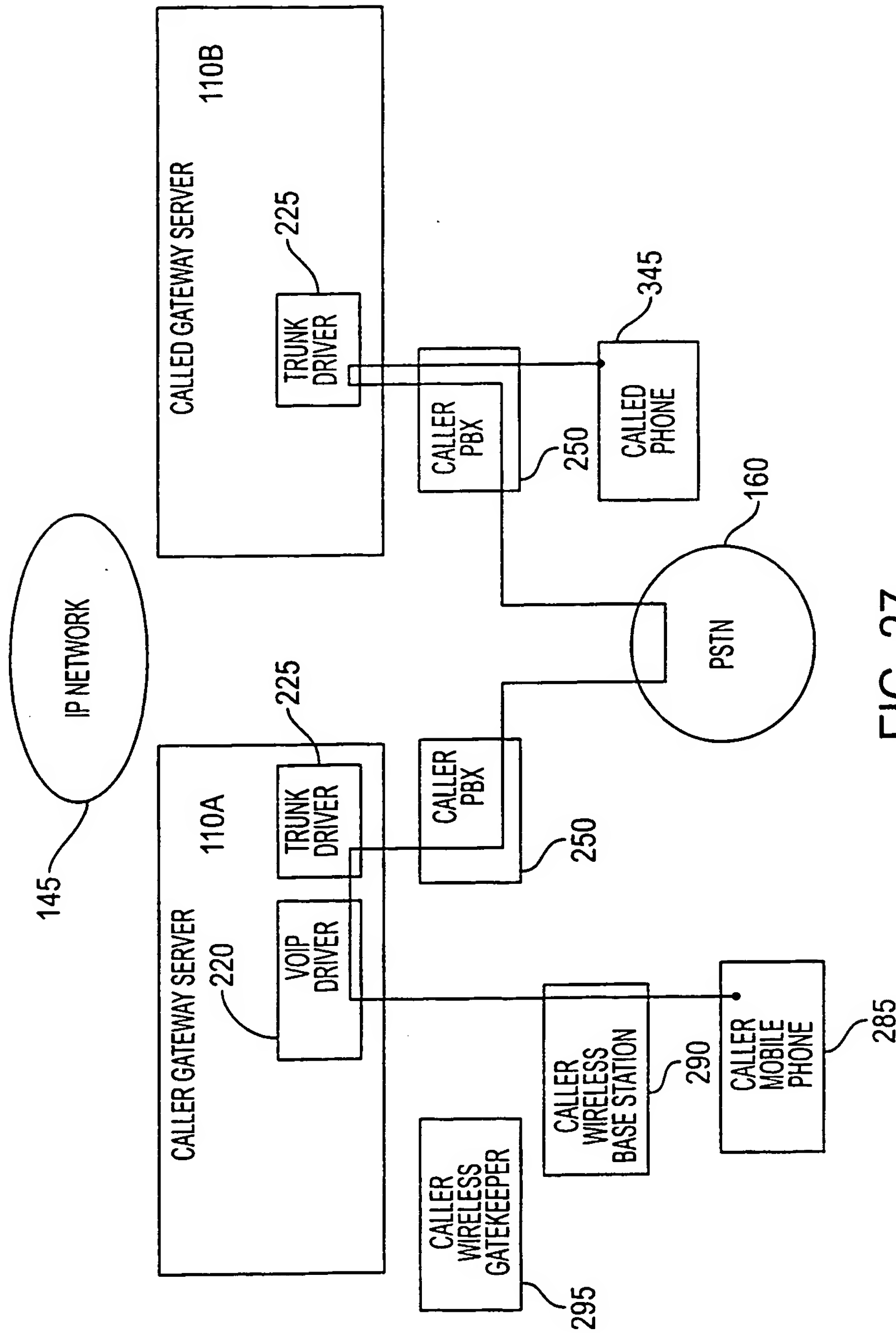

CALLED GATEWAY SERVER
110B
TRUNK DRIVER
225
CALLER PBX
250
CALLED PHONE
345
IP NETWORK
145
PSTN
160
CALLER GATEWAY SERVER
110A
TRUNK DRIVER
225
VOIP DRIVER
220
CALLER PBX
250
CALLER WIRELESS BASE STATION
290
CALLER MOBILE PHONE
285
CALLER WIRELESS GATEKEEPER
295

FIG. 27

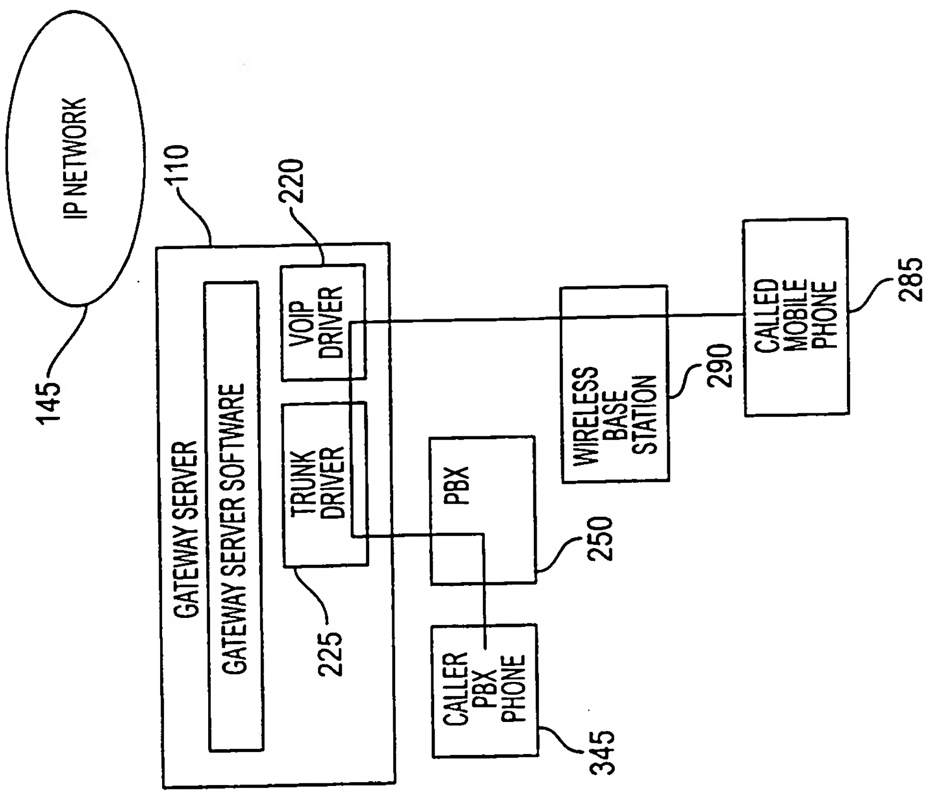


FIG. 28

BNSDOCID: <WO 0069156A1 I >

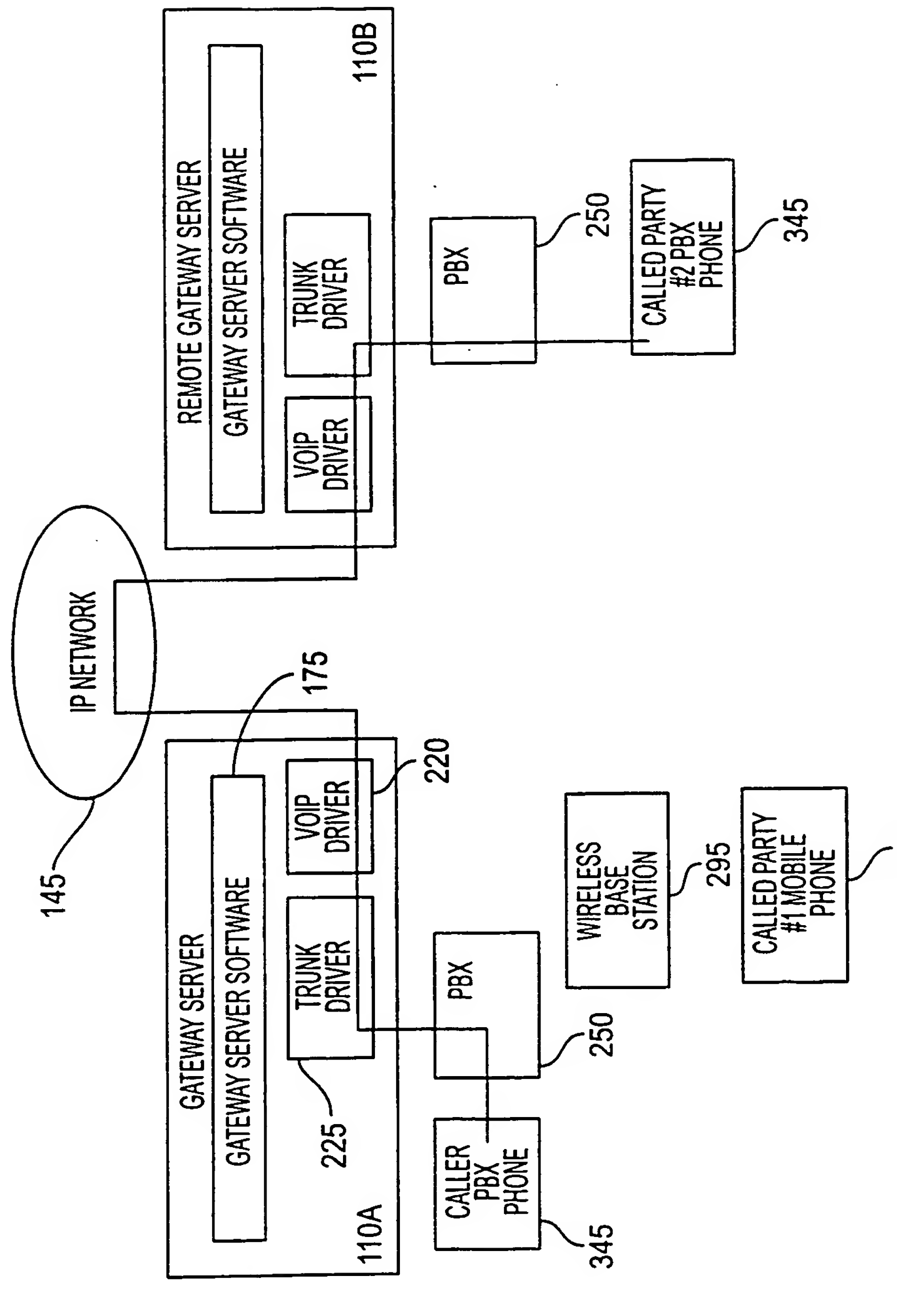

REMOTE GATEWAY SERVER
GATEWAY SERVER SOFTWARE
TRUNK DRIVER
VOIP DRIVER
110B
PBX
250
CALLED PARTY #2 PBX PHONE
345
IP NETWORK
145
175
GATEWAY SERVER
GATEWAY SERVER SOFTWARE
VOIP DRIVER
220
TRUNK DRIVER
225
110A
PBX
250
CALLER PBX PHONE
345
WIRELESS BASE STATION
295
CALLED PARTY #1 MOBILE PHONE
285

FIG. 29

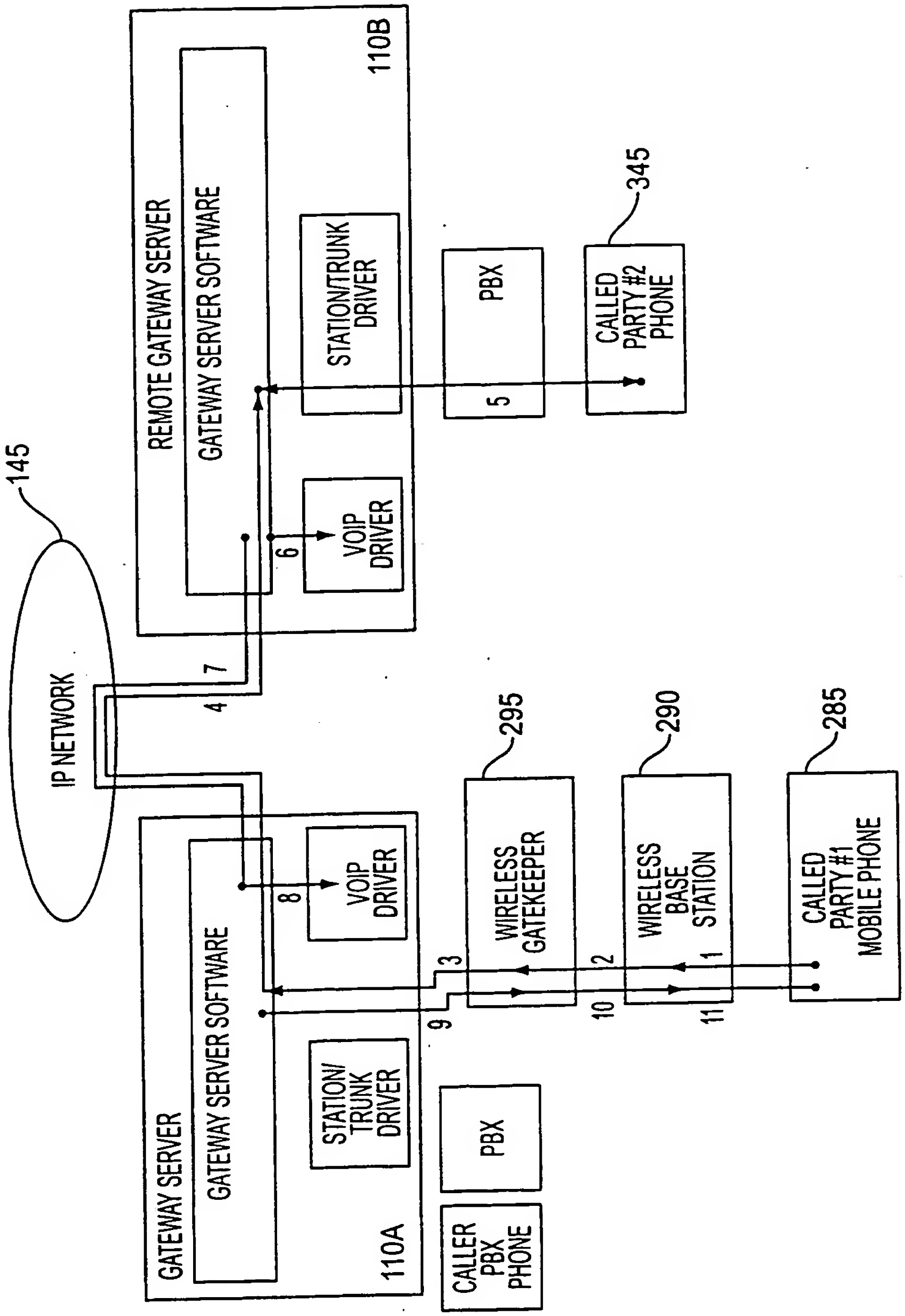

REMOTE GATEWAY SERVER
GATEWAY SERVER SOFTWARE
STATION/TRUNK DRIVER
VOIP DRIVER
110B
PBX
CALLED PARTY #2 PHONE
345
145
IP NETWORK
GATEWAY SERVER
GATEWAY SERVER SOFTWARE
VOIP DRIVER
STATION/ TRUNK DRIVER
110A
PBX
CALLER PBX PHONE
WIRELESS GATEKEEPER
295
WIRELESS BASE STATION
290
CALLED PARTY #1 MOBILE PHONE
285
1
2
3
4
5
6
7
8
9
10
11

FIG. 30

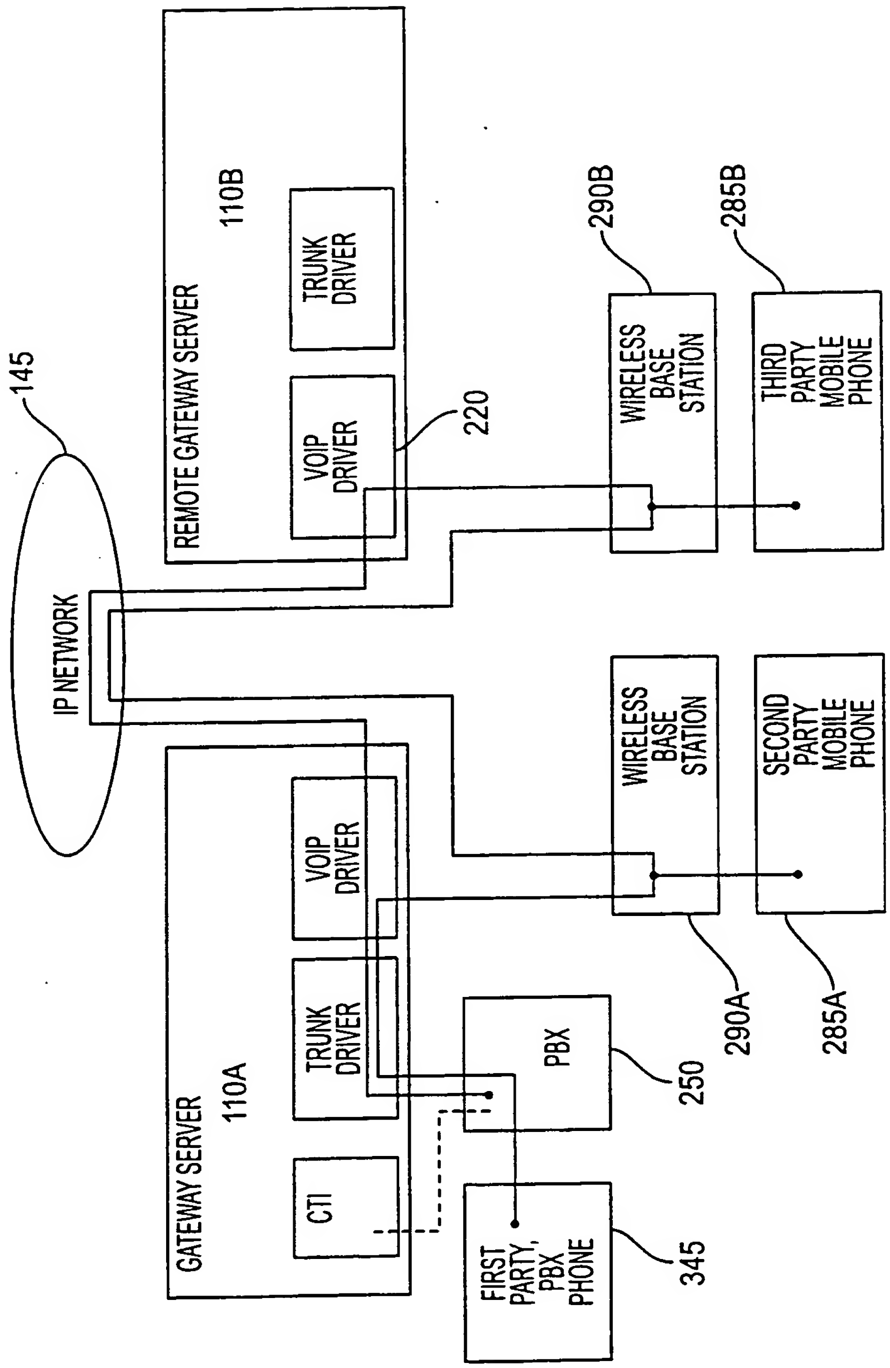

REMOTE GATEWAY SERVER
110B
TRUNK DRIVER
VOIP DRIVER
220
145
IP NETWORK
WIRELESS BASE STATION
290B
THIRD PARTY MOBILE PHONE
285B
GATEWAY SERVER
110A
CTI
TRUNK DRIVER
VOIP DRIVER
PBX
250
FIRST PARTY, PBX PHONE
345
WIRELESS BASE STATION
290A
SECOND PARTY MOBILE PHONE
285A

FIG. 31

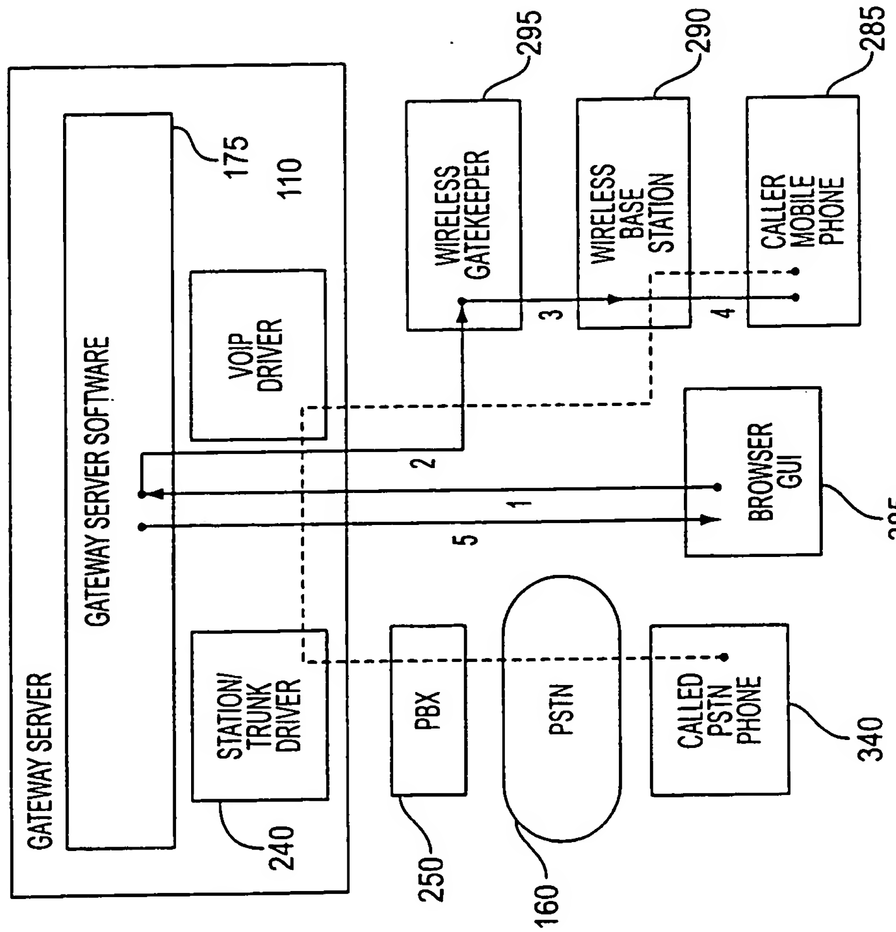

GATEWAY SERVER
GATEWAY SERVER SOFTWARE
175
110
VOIP DRIVER
STATION/ TRUNK DRIVER
240
WIRELESS GATEKEEPER
295
WIRELESS BASE STATION
290
CALLER MOBILE PHONE
285
BROWSER GUI
385
PBX
250
PSTN
160
CALLED PSTN PHONE
340
1
2
3
4
5

FIG. 32

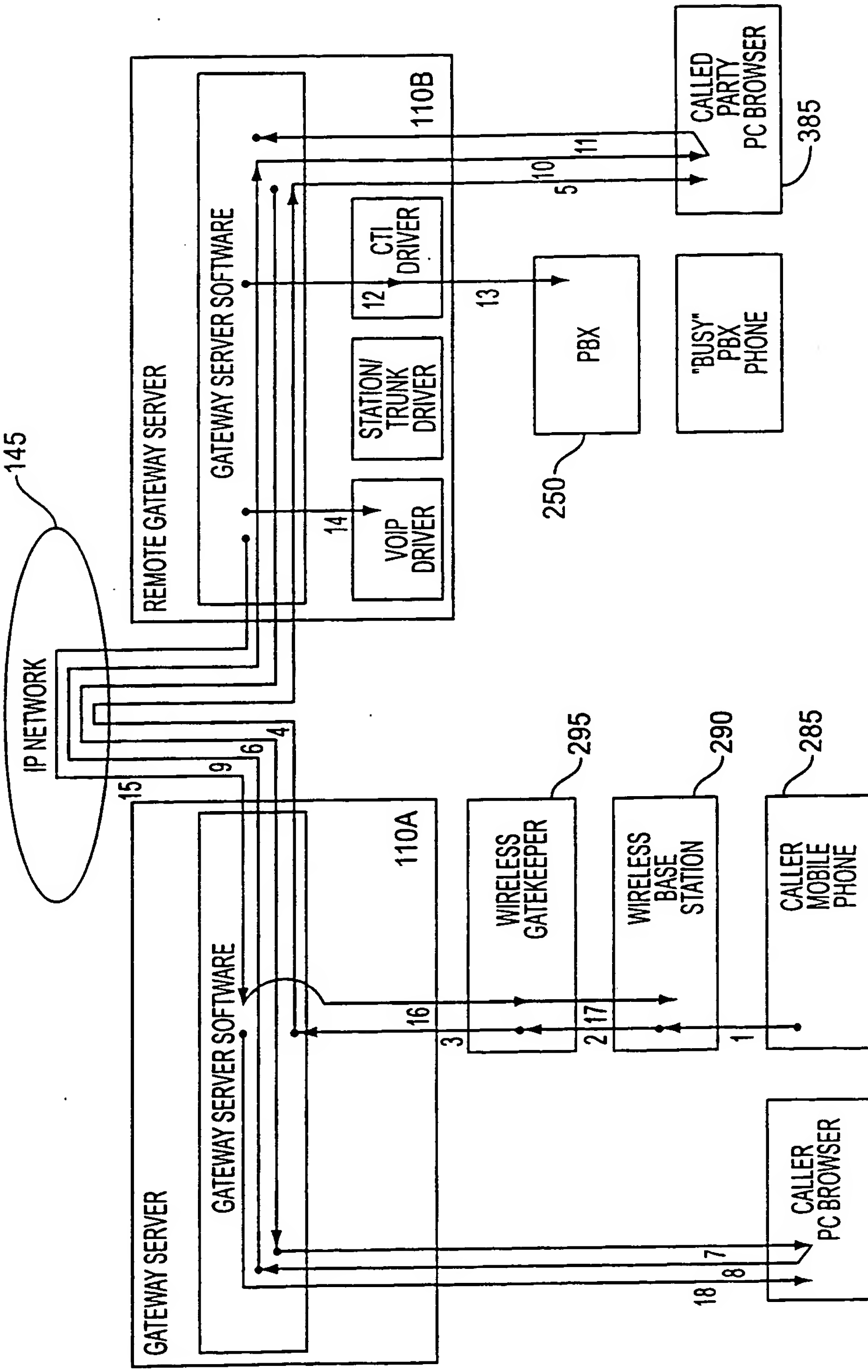

REMOTE GATEWAY SERVER
GATEWAY SERVER SOFTWARE
110B
VOIP DRIVER
STATION/ TRUNK DRIVER
CTI DRIVER
PBX
250
"BUSY" PBX PHONE
CALLED PARTY PC BROWSER
385
IP NETWORK
145
GATEWAY SERVER
GATEWAY SERVER SOFTWARE
110A
WIRELESS GATEKEEPER
295
WIRELESS BASE STATION
290
CALLER MOBILE PHONE
285
CALLER PC BROWSER
385
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16
17
18

FIG. 33

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 00/13247

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04M7/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04L H04M

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, PAJ, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 99 05590 A (HARBISON ROBERT W ;LO MING C (US); STARVOX INC (US); BARRY RICHARD) 4 February 1999 (1999-02-04) abstract page 3, line 30 -page 5, line 27	1,2, 4-17,20
Y	page 6, line 12 -page 7, line 17	3,19
P,X	TRICHT VAN E ET AL: "VOICE-OVER-IP FOR CORPORATE USERS. A SOLUTION IN SEARCH OF A PROBLEM?" GB,LONDON: IBTE, 24 August 1999 (1999-08-24), pages 9-14, XP000847162 page 9, middle column, line 29 -right-hand column, line 10 page 10, left-hand column, line 61 -right-hand column, line 9	1,13

-/--

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
13 October 2000	20/10/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Larcinese, C

1

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 00/13247

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	E MARGULIES: "Spawn of NT and ATM: The Un - PBX" COMPUTER TELEPHONY, US, SOUTHAMPTON, 1 November 1996 (1996-11-01), page 72,74,76,78 XP002079016 ISSN: 1072-1711 page 74, right-hand column, line 46 -page 76, right-hand column, line 9 ---	3,19
A	THOM G A: "H. 323: THE MULTIMEDIA COMMUNICATIONS STANDARD FOR LOCAL AREA NETWORKS" IEEE COMMUNICATIONS MAGAZINE, US, IEEE SERVICE CENTER. PISCATAWAY, N.J, vol. 34, no. 12, 1 December 1996 (1996-12-01), pages 52-56, XP000636454 ISSN: 0163-6804 the whole document -----	1-20

1

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/US 00/13247

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9905590 A	04-02-1999	AU 8576798 A	16-02-1999
		EP 1021757 A	26-07-2000

Form PCT/ISA/210 (patent family annex) (July 1992)

THIS PAGE BLANK (USPTO)

This Page is Inserted by IFW Indexing and Scanning Operations and is not part of the Official Record

BEST AVAILABLE IMAGES

Defective images within this document are accurate representations of the original documents submitted by the applicant.

Defects in the images include but are not limited to the items checked:

- ☐ BLACK BORDERS
- ☐ IMAGE CUT OFF AT TOP, BOTTOM OR SIDES
- ☐ FADED TEXT OR DRAWING
- ☐ BLURRED OR ILLEGIBLE TEXT OR DRAWING
- ☐ SKEWED/SLANTED IMAGES
- ☐ COLOR OR BLACK AND WHITE PHOTOGRAPHS
- ☐ GRAY SCALE DOCUMENTS
- ☑ LINES OR MARKS ON ORIGINAL DOCUMENT
- ☐ REFERENCE(S) OR EXHIBIT(S) SUBMITTED ARE POOR QUALITY
- ☐ OTHER: ________________________________

IMAGES ARE BEST AVAILABLE COPY.
As rescanning these documents will not correct the image problems checked, please do not report these problems to the IFW Image Problem Mailbox.